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UNITED STATES DISTRICT COURT

NORTHERN DISTRICT OF CALIFORNIA

SAN FRANCISCO DIVISION

REALNETWORKS, INC., Plaintiff, v. BURST.COM, INC., Defendant.	Case No. 08-CV-0023 MHP **FIRST AMENDED COMPLAINT FOR DECLARATORY JUDGMENT** **DEMAND FOR JURY TRIAL** ***(CORRECTED VERSION)***

Plaintiff RealNetworks, Inc. ("RealNetworks") for its First Amended Complaint for Declaratory Judgment against Defendant Burst.com, Inc. ("Burst.com") alleges as follows:

1. This is a civil action arising under the Patent Laws of the United States, 35 U.S.C. §§ 101, *et seq.*, seeking a declaratory judgment that United States Patent No. 4,963,995 ("'995

patent"), United States Patent No. 5,164,839 ("'839 patent"), United States Patent No. 5,995,705 ("'705 patent"), United States Patent No. 5,057,932 ("'932 patent"), United States Patent No. 5,963,202 ("'202 patent"), United States Patent No. 6,850,965 ("'965 patent") and United States Patent No. 7,334,044 ("'044 patent") (collectively "patents-in-suit") are invalid and/or unenforceable and not infringed by Plaintiff RealNetworks, Inc.

PARTIES

2. Plaintiff RealNetworks is a corporation organized and existing under the laws of the State of Washington, with its principal place of business at 2601 Elliott Avenue, Seattle, Washington.

3. Defendant Burst.com, Inc., previously known as Instant Video Technologies, Inc., is a corporation organized and existing under the laws of the State of Delaware and appears or represents itself to be the assignee of record of the patents-in-suit. On information and belief, Burst.com currently maintains its principal place of business at 613 Fourth St., Suite 201, Santa Rosa, California.

4. On information and belief, Burst.com owns and licenses patents and does not make or sell any products or services. Richard Lang, Burst.com's CEO, is the named inventor listed on four of the patents-in-suit.

5. Three of the seven patents-in-suit were the subject of an earlier action captioned *Burst.com, Inc. v. Microsoft Corp.*, Civ. No. JFM-02-2952, in the District of Maryland ("Microsoft Litigation"). The Microsoft Litigation concluded by settlement on or about March 11, 2005. Four of the seven patents-in-suit, the '995, '839, '705, and '932 patents, were also the subject of another action, *Apple Computer, Inc. v. Burst.com, Inc.*, Case No. C06-00019 MHP, in the Northern District of California ("Apple Litigation"). The Apple Litigation was dismissed pursuant to settlement on or about December 4, 2007.

JURISDICTION AND VENUE

6. RealNetworks brings this Complaint against Burst.com pursuant to the Patent Laws of the United States, Title 35 of the United States Code, with a specific remedy sought based upon the laws authorizing actions for declaratory judgment in the courts of the United States, 28 U.S.C. §§ 2201 and 2202.

7. This Court has subject matter jurisdiction over this action, which arises under the patent laws of the United States, pursuant to 28 U.S.C. §§ 1331, 1338(a), and 2201.

8. Venue in this district is proper under 28 U.S.C. §§ 1391(b) and (c) and 1400(b) because, on information and belief, Burst.com's principal place of business is in this district, Burst.com is subject to personal jurisdiction in this district, and Burst.com previously asserted the patents-in-suit against Microsoft Corp. and Apple Computer, Inc. in this district.

EXISTENCE OF AN ACTUAL CONTROVERSY

9. There is an actual controversy within the jurisdiction of this Court under 28 U.S.C. §§ 2201 and 2202.

10. Burst.com approached RealNetworks in or about November 18, 1997, within two months after RealNetworks filed its S-1 Registration Statement, demanding that RealNetworks obtain a license to the patents owned by Burst.com. Burst.com identified certain patents it claimed to own, including several of the patents-in-suit, in a letter to RealNetworks from its attorneys, Carr & Ferrell LLP.

11. On information and belief, Carr & Ferrell LLP prosecuted the '705 patent.

12. On or about April 9, 1998, RealNetworks informed Burst.com that it did not believe that any RealNetworks products were covered by the '995, '839, '705, and '932 patents. In response, Burst.com stated that it "reserves its rights to proceed with an infringement action against RealNetworks."

13. The Microsoft Litigation commenced on or about June 18, 2002 and concluded by settlement on or about April 15, 2005. Under the terms of the settlement of the Microsoft Litigation, Microsoft agreed to pay $60 million to Burst.com and was granted a license to Burst.com's entire patent portfolio.

14. On information and belief, Microsoft was Burst.com's sole patent licensee and its sole source of licensing revenue until the recent settlement of the Apple Litigation pursuant to which Apple reportedly paid Burst.com $10 million.

15. In late 2005, Burst.com again threatened litigation against RealNetworks. Following extensive discussions, Burst.com and RealNetworks entered into a Standstill Agreement whereby the parties agreed not to commence litigation against the other relating to Burst.com's patents through October 31, 2005 ("Standstill Agreement").

16. Burst.com and RealNetworks thereafter agreed to several extensions of the Standstill Agreement. Pursuant to its terms, the most recent Standstill Agreement, entered into on or about June 16, 2006, was scheduled to expire thirty days after the termination of the Apple Litigation.

17. The Apple Litigation, which commenced on or about January 4, 2006, was terminated on or about December 4, 2007 pursuant to settlement. Accordingly, the Standstill Agreement between Burst.com and RealNetworks expired as of January 3, 2008.

18. On January 8, 2008, Burst.com advised RealNetworks that Burst.com is accusing RealNetworks of infringing the '202 patent.

19. On January 30, 2008 Burst.com advised RealNetworks that Burst.com is accusing RealNetworks of infringing the '965 patent, as well as claims included in the '044 patent, which the United States Patent and Trademark Office ("PTO") issued on February 19, 2008.

20. RealNetworks denies that any of the patents-in-suit are or have been infringed by RealNetworks and disputes their validity and enforceability.

21. Based on Burst.com's extensive communications with RealNetworks, its assertions that RealNetworks infringes the patents-in-suit, and its threats that it would engage in litigation with RealNetworks, all directed at RealNetworks, Burst.com has created in RealNetworks a reasonable apprehension that it will initiate a patent infringement suit against RealNetworks, alleging that RealNetworks infringes the patents-in-suit.

22. An actual and justiciable controversy exists between RealNetworks and Burst.com as to whether the patents-in-suit are infringed, invalid, and/or unenforceable. Absent a declaration of noninfringement, invalidity and/or unenforceability, Burst.com will continue to wrongfully assert the patents-in-suit against RealNetworks, and thereby cause RealNetworks irreparable injury and damage.

COUNT I:
Declaratory Judgment of Noninfringement of '995 Patent

23. RealNetworks hereby restates and realleges the allegations set forth in paragraphs 1 through 22 above and incorporates them by reference.

24. RealNetworks has not infringed, and is not infringing, either directly or indirectly, contributorily or otherwise, any claim of the '995 patent. RealNetworks has not induced others to infringe the '995 patent. A true and correct copy of the '995 patent is attached as Exhibit A.

25. RealNetworks is entitled to a declaratory judgment that it does not infringe the '995 patent.

COUNT II:
Declaratory Judgment of Invalidity of '995 Patent

26. RealNetworks hereby restates and realleges the allegations set forth in paragraphs 1 through 25 above and incorporates them by reference.

27. The claims of the ’995 patent are invalid for failure to comply with the requirements of the Patent Laws of the United States, including without limitation 35 U.S.C. §§ 101, 102, 103 and 112.

28. RealNetworks is entitled to a declaratory judgment that the ’995 patent claims are invalid.

COUNT III:
Declaratory Judgment of Noninfringement of ’839 Patent

29. RealNetworks hereby restates and realleges the allegations set forth in paragraphs 1 through 28 above and incorporates them by reference.

30. RealNetworks has not infringed, and is not infringing, either directly or indirectly, contributorily or otherwise, any claim of the ’839 patent. RealNetworks has not induced others to infringe the ’839 patent. A true and correct copy of the ’839 patent is attached as Exhibit B.

31. RealNetworks is entitled to a declaratory judgment that it does not infringe the ’839 patent.

COUNT IV:
Declaratory Judgment of Invalidity of ’839 Patent

32. RealNetworks hereby restates and realleges the allegations set forth in paragraphs 1 through 31 above and incorporates them by reference.

33. The claims of the ’839 patent are invalid for failure to comply with the requirements of the Patent Laws of the United States, including without limitation 35 U.S.C. §§ 101, 102, 103 and 112.

34. RealNetworks is entitled to a declaratory judgment that the ’839 patent claims are invalid.

<u>COUNT V</u>:
Declaratory Judgment of Noninfringement of ’705 Patent

35. RealNetworks hereby restates and realleges the allegations set forth in paragraphs 1 through 34 above and incorporates them by reference.

36. RealNetworks has not infringed, and is not infringing, either directly or indirectly, contributorily or otherwise, any claim of the ’705 patent. RealNetworks has not induced others to infringe the ’705 patent. A true and correct copy of the ’705 patent is attached as Exhibit C.

37. RealNetworks is entitled to a declaratory judgment that it does not infringe the ’705 patent.

<u>COUNT VI</u>:
Declaratory Judgment of Invalidity of ’705 Patent

38. RealNetworks hereby restates and realleges the allegations set forth in paragraphs 1 through 37 above and incorporates them by reference.

39. The claims of the ’705 patent are invalid for failure to comply with the requirements of the Patent Laws of the United States, including without limitation 35 U.S.C. §§ 101, 102, 103 and 112.

40. RealNetworks is entitled to a declaratory judgment that the ’705 patent claims are invalid.

<u>COUNT VII</u>:
Declaratory Judgment of Noninfringement of ’932 Patent

41. RealNetworks hereby restates and realleges the allegations set forth in paragraphs 1 through 40 above and incorporates them by reference.

42. RealNetworks has not infringed, and is not infringing, either directly or indirectly, contributorily or otherwise, any claim of the ’932 patent. RealNetworks has not induced others to infringe the ’932 patent. A true and correct copy of the ’932 patent is attached as Exhibit D.

43. RealNetworks is entitled to a declaratory judgment that it does not infringe the ’932 patent.

<u>COUNT VIII</u>:
Declaratory Judgment of Invalidity of ’932 Patent

44. RealNetworks hereby restates and realleges the allegations set forth in paragraphs 1 through 43 above and incorporates them by reference.

45. The claims of the ’932 patent are invalid for failure to comply with the requirements of the Patent Laws of the United States, including without limitation 35 U.S.C. §§ 101, 102, 103 and 112.

46. RealNetworks is entitled to a declaratory judgment that the ’932 patent claims are invalid.

<u>COUNT IX</u>:
Declaratory Judgment of Noninfringement of ’202 Patent

47. RealNetworks hereby restates and realleges the allegations set forth in paragraphs 1 through 46 above and incorporates them by reference.

48. RealNetworks has not infringed, and is not infringing, either directly or indirectly, contributorily or otherwise, any claim of the ’202 patent. RealNetworks has not induced others to infringe the ’202 patent. A true and correct copy of the ’202 patent is attached as Exhibit E.

49. RealNetworks is entitled to a declaratory judgment that it does not infringe the ’202 patent.

<u>COUNT X</u>:
Declaratory Judgment of Invalidity of ’202 Patent

50. RealNetworks hereby restates and realleges the allegations set forth in paragraphs 1 through 49 above and incorporates them by reference.

51. The claims of the ’202 patent are invalid for failure to comply with the requirements of the Patent Laws of the United States, including without limitation 35 U.S.C. §§ 101, 102, 103 and 112.

52. RealNetworks is entitled to a declaratory judgment that the ’202 patent claims are invalid.

COUNT XI:
Declaratory Judgment of Noninfringement of ’965 Patent

53. RealNetworks hereby restates and realleges the allegations set forth in paragraphs 1 through 52 above and incorporates them by reference.

54. RealNetworks has not infringed, and is not infringing, either directly or indirectly, contributorily or otherwise, any claim of the ’965 patent. RealNetworks has not induced others to infringe the ’965 patent. A true and correct copy of the ’965 patent is attached as Exhibit F.

55. RealNetworks is entitled to a declaratory judgment that it does not infringe the ’965 patent.

COUNT XII
Declaratory Judgment of Invalidity of ’965 Patent

56. RealNetworks hereby restates and realleges the allegations set forth in paragraphs 1 through 55 above and incorporates them by reference.

57. The claims of the ’965 patent are invalid for failure to comply with the requirements of the Patent Laws of the United States, including without limitation 35 U.S.C. §§ 101, 102, 103 and 112.

58. RealNetworks is entitled to a declaratory judgment that the ’965 patent claims are invalid.

<u>COUNT XIII</u>:
Declaratory Judgment of Noninfringement of ’044 Patent

59. RealNetworks hereby restates and realleges the allegations set forth in paragraphs 1 through 58 above and incorporates them by reference.

60. RealNetworks has not infringed, and is not infringing, either directly or indirectly, contributorily or otherwise, any claim of the ’044 patent. RealNetworks has not induced others to infringe the ’965 patent. A true and correct copy of the ’044 patent is attached as Exhibit G.

61. RealNetworks is entitled to a declaratory judgment that it does not infringe the ’044 patent.

<u>COUNT XIV:</u>
Declaratory Judgment of Invalidity of ’044 Patent

62. RealNetworks hereby restates and realleges the allegations set forth in paragraphs 1 through 61 above and incorporates them by reference.

63. The claims of the ’044 patent are invalid for failure to comply with the requirements of the Patent Laws of the United States, including without limitation 35 U.S.C. §§ 101, 102, 103 and 112.

64. RealNetworks is entitled to a declaratory judgment that the ’044 patent claims are invalid.

<u>COUNT XV</u>:
Declaratory Judgment of Unenforceability of the ‘995, ‘839, ‘705, and ‘932 Patents

65. RealNetworks hereby restates and realleges the allegations set forth in paragraphs 1 through 64 above and incorporates them by reference.

66. Burst.com's allegation that RealNetworks infringes the '995, '839, '705, and '932 patents is barred because these patents are unenforceable pursuant to 37 C.F.R. § 1.56 and the doctrine of inequitable conduct.

67. On information and belief, Richard Lang, Burst.com's CEO and named inventor on four of the patents-in-suit, and/or the prosecuting attorneys of the '995 patent committed inequitable conduct during prosecution of the patents-in-suit by intentionally withholding from the PTO the fact that Mr. Lang was incapable of practicing the claimed invention, and by disclosing only an inoperable embodiment of the claimed invention instead of making an enabling disclosure. Specifically, Lang disclosed only one device for performing compression in the '995 patent, the AMD 7971 fax compression chip. This chip could not be used to practice the claimed invention using the disclosure in Lang's application. Further, Lang knew that the only disclosed embodiment of the '995 patent was inoperable and would not enable one of skill in the art to practice the invention. Lang himself could not have built the described embodiment using the disclosure in the patent. Further, at the time of filing the application that led to the '995 patent, and also at the time of filing the application that led to the '932 patent, Lang was not aware of any existing technology that was actually capable of performing the compression described in his application, and believed instead that the AMD 7971 fax chip was the best mode of performing video compression in his invention. At no time did Lang and/or the prosecuting attorneys of the '995 patent disclose or explain to the PTO that the only disclosed embodiment of the '995 patent was inoperable as described, that the disclosure was not enabling, or that Lang could not practice the invention. The failure to disclose this material information was knowing, willful and done with the intent to deceive the PTO into issuing the '995 patent.

68. On information and belief, in the continuation-in-part application that led to the '932, '839, and '705 patents, Lang and/or the prosecuting attorneys removed all reference to the AMD 7971 chip because they knew it did not work. By stripping out all reference to the AMD

7971 chip, which Lang believed to be the best mode of the invention, Lang intentionally withheld the best mode of the invention from the ’932, ’839, and ’705 patents. At no time did Lang disclose or explain to the PTO that the best mode was not disclosed in the ’932, ’839, and ’705 patents.

69. A reasonable examiner would have found it important to know that the only disclosed embodiment of the ’995 patent was inoperable as described, that the disclosure was not enabling, that Mr. Lang could not practice the invention, and that Mr. Lang’s best mode had been removed from the continuation-in-part application that led to the ’932, ’839, and ’705 patents. The failure to disclose this material information was knowing, willful and done with the intent to deceive the PTO into issuing the ’932, ’839, ’705 patents. As a result, the ’932, ’839, and ’705 patents are unenforceable.

70. On information and belief, at the time of filing the application that led to the ’995 patent and also at the time of filing the application that led to the ’932 patent, a continuation-in-part application that depends from the ’995 patent, Lang and/or the prosecuting attorneys were unaware of any existing technology that was actually capable of performing the compression described in those applications but failed to disclose this material information to the PTO. The failure to disclose this material information was knowing, willful and done with the intent to deceive the PTO into issuing the ’995 and ’932 patents. As a result, the ’995 and ’932 patents are unenforceable.

71. In an Office Action dated April 22, 1994, the European Patent Office (“EPO”) rejected the then pending claims of Burst.com’s Application No. 90 902 741.9 based on, among other grounds, that the claims “introduce subject matter which extends beyond the content of the application as originally filed.” The EPO noted that while the claims required a “time compressed representation” and “storing of time compressed representation,” “no such time compression or storing of time compressed information could be identified” in the application.

72. In response, Burst.com argued on May 5, 1995 that the existing descriptions disclosed that "the video program is communicated or transmitted at an accelerated rate in less time that would be taken to view the program" and thus that no new subject matter had been introduced by "time compressed representation" in the claims. The EPO responded on June 30, 1995, that the passage cited by Burst.com did not disclose time compression, but disclosed only data compression and faster-than real-time transmission. The EPO noted that claims that involved storing a time compressed representation did not make sense: "the subject matter as claimed in claims 1, 40, 50, 66, and 105 lacks clarity because it does not make sense to generate a time compressed representation of an information and store this representation in a memory means where the effect of the time compression is lost." In response, Burst.com deleted "time" from its claims and claimed only compression plus sending the compressed representation faster than real time. Burst.com also amended the description to match the new claims.

73. On information and belief, Lang and/or the prosecuting attorneys of the patents in suit were aware of the rejections in the EPO because they were directly involved in the prosecution. Lang and/or the prosecuting attorneys never informed the PTO that Burst.com's European claims including "time compressed representation" had been rejected by the EPO for lack of support in the written description, despite the fact that its U.S. claims were purportedly based on the same description. The EPO rejections of the "time compressed representation" language would have been important to a reasonable examiner in deciding whether to allow the Burst.com applications to issue in the U.S. The EPO rejections and Burst.com's response to them would have been particularly material in light of statements made by Burst.com to the PTO that contradicted the substance of Burst.com's dialog with the EPO. For example, Burst.com stated during prosecution of the '705 patent that "time compression" is "defined in the specification of the Application" when trying to traverse a rejection over the prior art. Burst.com's withholding from the PTO of EPO rejections based on the fact that "time compression" is not supported by the

written description, as well of its actions in response, was knowing, willful and done with the intent to deceive the PTO into issuing the ’705 patent. As a result, the ’705 patent is unenforceable.

74. On information and belief, Lang and/or the prosecuting attorneys of the patents-in-suit intentionally misrepresented the content of U.S. Patent No. 4,506,387 by Howard Walter (“Walter Patent”) to the examiner in a manner calculated to conceal the significance of Walter Patent from the examiner. The Walter Patent was first disclosed to the PTO in an Information Disclosure Statement (“IDS”) on May 6, 1991. In that IDS, the applicant stated that the Walter Patent had first come to his attention on April 2, 1991, was discussed with his attorney, on April 9, 1991, and that they had concluded the Walter Patent was material to the examination of the application that led to the ’932 patent. The applicant described the Walter Patent as being “directed to a programming-on-demand cable TV system in which a video program is divided into a number of segments and stored in a segmented memory in compressed digital form. The stored video segments are then converted from electrical data to optical data and simultaneously transmitted over a plurality of parallel fiber optic transmission lines to a data receiving station, which then reconverts the optical data back to the original electrical data.” Applicant’s description of the Walter Patent was intentionally misleading and deceptive because it suppressed the most material facts about the reference, including that the Walter Patent teaches data compression that enables significantly faster-than-real-time transmission of compressed full motion video. A reasonable U.S. patent examiner would have at all times considered the fact that the Walter Patent disclosed faster-than-real-time transmission of compressed video important in deciding whether to allow the applications for the Burst.com patents to issue.

75. On information and belief, Lang suppressed the material facts about the Walter Patent with intent to deceive the PTO. Lang knew that the Walter Patent disclosed faster-than-real-time transmission of data-compressed video. Before disclosing the Walter Patent to the PTO,

Lang had studied the Walter Patent and its file history on several occasions and had concluded that it was material prior art. Despite this extensive review, Lang and/or the prosecuting attorneys summarized the disclosure of Walter in a misleading and deceptive way to hide its true importance from the PTO. The suppression of this information was knowing, willful and done with the intent to deceive the PTO. As a result, the ’995, ’932, ’839, and ’705 patents are unenforceable.

76. On information and belief, Mr. Lang and/or the prosecuting attorneys of the ’995, ’932, ’839, and ’705 patents withheld from the PTO critical information regarding DVI compression technology. The DVI compression technology was a revolutionary advance in video compression that allowed full motion video to be played and stored at the CD-ROM data rate of 150 KB/second. The DVI compression technology was developed during the mid-1980s at the David Sarnoff Research Center in Princeton, NJ. DVI was demonstrated publicly numerous times before the filing of the Burst.com patents, including in March 1987 at the Microsoft CD-ROM conference. DVI and its public demonstrations were also detailed in numerous printed publications before the filing of the Burst.com patents.

77. On information and belief, Lang learned about DVI no later than September 1990. At that time, Burst.com was attempting to build a prototype of its system, but had been unable to develop a workable compression technique. Burst.com learned that DVI, now owned by Intel, had already succeeded in compressing video to a CD-ROM data rate. Burst.com abandoned its efforts to develop its own compression techniques, and adopted DVI technology for use in Burst.com prototypes.

78. The DVI system could take uncompressed full motion video and dramatically reduce the data rate of the signal so as to allow storage, faster-than-real-time transfer, and real-time playing of the compressed video signal. On information and belief, Lang knew this because he used DVI to do exactly that in the prototype of his claimed invention exhibited at CES in 1991.

DVI was developed and publicized in the United States years before the Burst.com patents were filed. DVI is material to the patentability of the claims of the ’995, ’932, ’839, and ’705 patents, and a reasonable patent examiner would have considered DVI important in deciding whether to allow the claims of the ’995, ’932, ’839, and ’705 patents.

79. On information and belief, despite Burst.com’s intimate knowledge of DVI and its own use of the technology in its attempts to reduce Burst.com’s putative invention to practice, Lang and/or the prosecuting attorneys of the patents-in-suit withheld information about DVI from the PTO. Lang and/or the prosecuting attorneys failed to disclose printed publications describing DVI technology that was in development and displayed to the public before the filing of the Burst.com patents. The suppression of this information was knowing, willful and done with the intent to deceive the PTO. As a result, the ’995, ’932, ’839, and ’705 patents are unenforceable.

80. On information and belief, Lang and/or the prosecuting attorneys of the ’705 patent also intentionally withheld from the PTO prior art cited against the European equivalent of the Burst.com patents during prosecution of the ’705 patent. In an Office Action dated April 22, 1994 for Application No. 90 902 741.9, the European Patent Office (“EPO”) cited and discussed prior art references including IEEE Transactions on Consumer Electronics, “1988 International Conference on Consumer Electronics, Part 1”, 34 (1988) August, No. 3, New York, U.S., pages 838-845; Hildering et al.: “Programmable Compact Disk Picture Memory and Video Processing System” (“Hildering”); EP-A-0 283 727 (“Parker”); and EP-A-0 082 077 (“Gremillet EP”). The Hildering, Parker, and Gremillet EP references disclose subject matter material to the patentability of the asserted claims of the ’705 patent. In the EPO Office Action, all of the then-pending claims, which included the phrase “time compressed representation,” were deemed not patentable over at least one of the three references. A reasonable examiner would have at all times considered Hildering, Parker, and Gremillet EP important in deciding whether to allow the applications for the Burst.com patents to issue. Lang and/or the prosecuting attorney of the ’705

patent, who had knowledge of these references due to the EPO Office Action rejecting the claims in Burst.com's European application, withheld Hildering, Parker, and Gremillet EP from the PTO with the intent to deceive the PTO. As a result, the '705 patent is unenforceable.

81. Based on the above described intentional acts by Lang and/or the prosecuting attorneys of the patents-in-suit, all claims in the '995, '932, '839, and '705 patents are unenforceable due to inequitable conduct.

82. Furthermore, inequitable conduct during prosecution of any one of the Burst.com patents also renders unenforceable the claims of the other Burst.com patents by the doctrine of infectious unenforceability. The '932, '839, and '705 patents share the same specification. All three descend from the '995 parent patent. Specifically, the '705 patent is the ultimate continuation of the '839 patent, which itself is a division of a continuation-in-part of the '995 patent. That continuation-in-part of the '995 patent led to the '932 patent. Due to the immediate and necessary relationship the '995, '932, '839, and '705 patents have with each other, inequitable conduct during the prosecution of the any one of the Burst.com patents renders the others unenforceable as well.

PRAYER FOR RELIEF

WHEREFORE, RealNetworks prays for judgment as follows:

1. Declaring that RealNetworks has not infringed and is not infringing, directly, contributorily, or through inducement, any claims of the patents-in-suit;

2. Declaring that each of the patents-in-suit is invalid;

3. Declaring that each of the patents-in-suit is unenforceable;

4. Entering judgment in favor of RealNetworks and against Burst.com;

5. Declaring that Burst.com and each of their officers, employees, agents, alter egos, attorneys, and any persons in active concert or participation with them be restrained and enjoined from further prosecuting or instituting any action against RealNetworks claiming that the patents-

in-suit are valid or infringed, or from representing that any of RealNetworks' products or services, or others' use thereof, infringes the patents-in-suit;

6. Declaring this case exceptional under 35 U.S.C. § 285 and awarding RealNetworks its attorneys' fees and costs in connection with this case; and

7. Awarding RealNetworks such other and further relief as the Court deems just and proper.

DATED: February 20, 2008

By: /s/ Robert F. Kramer

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DEMAND FOR JURY TRIAL

Pursuant to the provisions of Rule 38(b) of the Federal Rules of Civil Procedure and Civil L.R. 3-6(a), Plaintiff RealNetworks, Inc. demands a trial by jury of all issues so triable in this matter.

DATED: February 20, 2008

By: /s/ Robert F. Kramer

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Attorneys for Plaintiff
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EXHIBIT A

United States Patent [19]

Lang

[11] Patent Number: 4,963,995

[45] Date of Patent: Oct. 16, 1990

[54] AUDIO/VIDEO TRANSCEIVER APPARATUS INCLUDING COMPRESSION MEANS

[75] Inventor: Richard A. Lang, Cave Creek, Ariz.

[73] Assignee: Explore Technology, Inc., Scottsdale, Ariz.

[21] Appl. No.: 289,776

[22] Filed: Dec. 27, 1988

[51] Int. Cl.[5] H04N 5/76

[52] U.S. Cl. 358/335; 358/133; 360/8; 360/9.1; 360/14.1

[58] Field of Search 360/9.1, 55, 13, 14.1, 360/8; 358/335, 903, 901, 133

[56] References Cited

U.S. PATENT DOCUMENTS

4,179,709	12/1979	Workman	358/133
4,446,490	5/1984	Hoshimi	360/13
4,511,934	4/1985	Ohira	360/55
4,516,156	5/1985	Fabris et al.	358/85
4,563,710	1/1986	Baldwin	360/9.1
4,625,080	11/1986	Scott	360/33.1
4,698,664	10/1987	Nichols	360/14.1
4,750,034	6/1988	Lem	358/335
4,768,110	8/1988	Dunlap et al.	360/33.1
4,774,574	9/1988	Daly et al.	358/133
4,851,931	7/1989	Parker et al.	360/115

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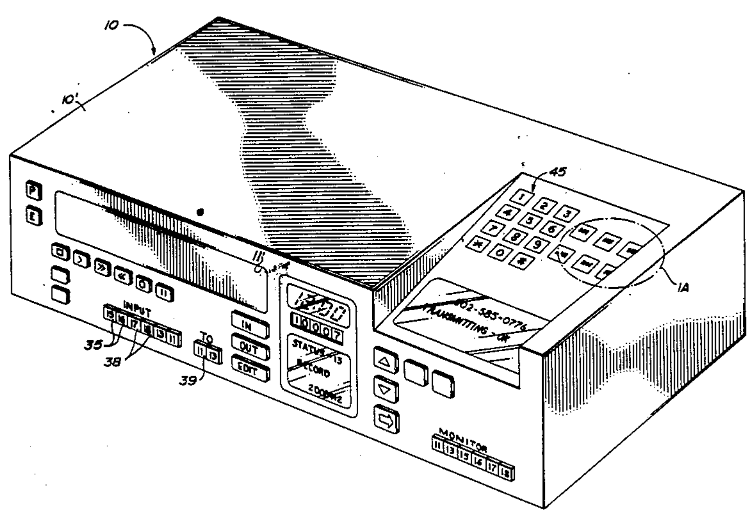
Attorney, Agent, or Firm—William E. Hein

[57] ABSTRACT

An improved video recorder/transmitter with expanded functionality including a capability for editing and/or copying from one video tape to another using only a single tape deck. The increased functionality is realized through the use of analog to digital conversion, signal compression and intermediate storage in an integrated circuit, random access memory. The recorder/-transmitter has capabilities to transmit and receive program information in either a compressed or decompressed format over fiber optic lines.

80 Claims, 2 Drawing Sheets



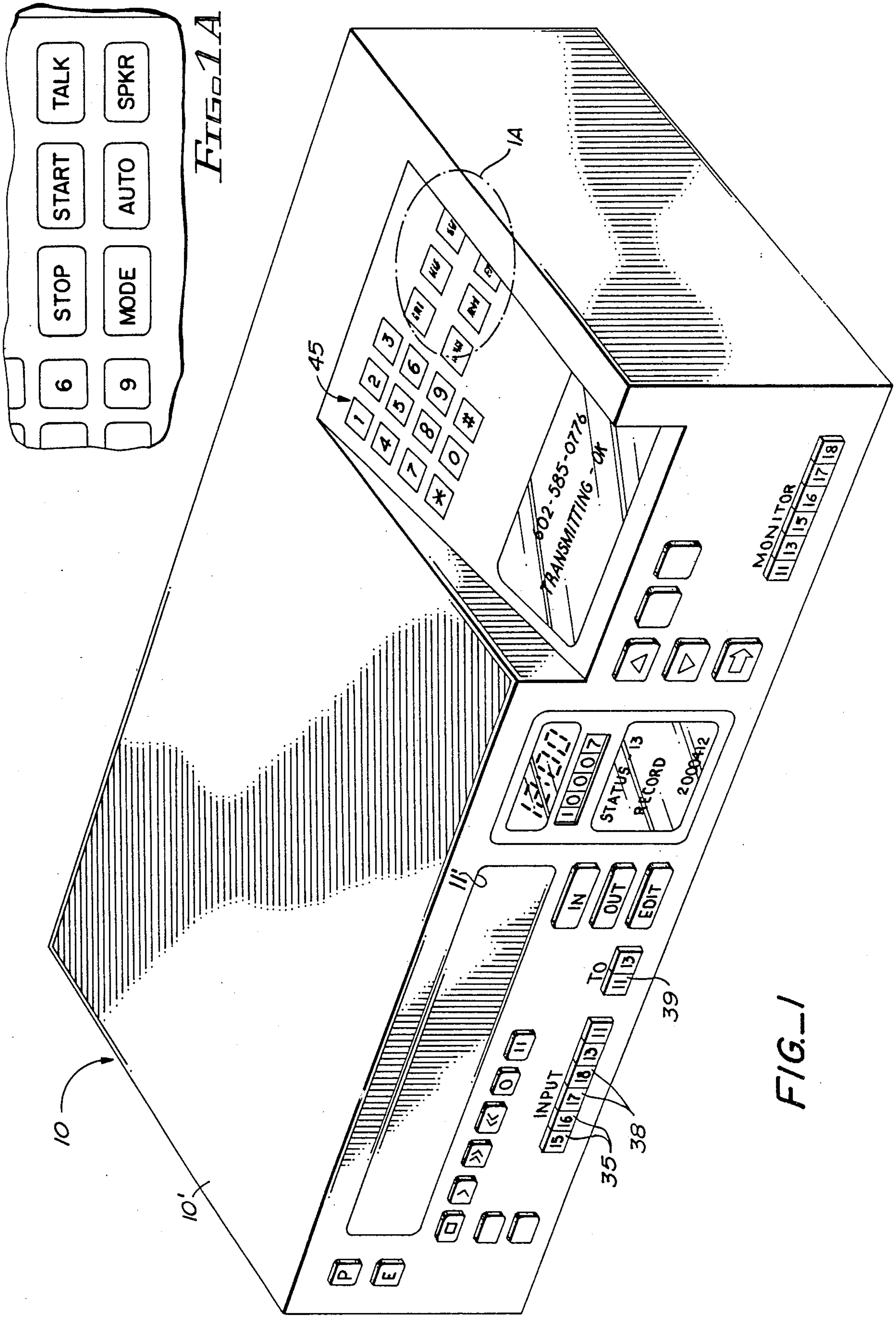

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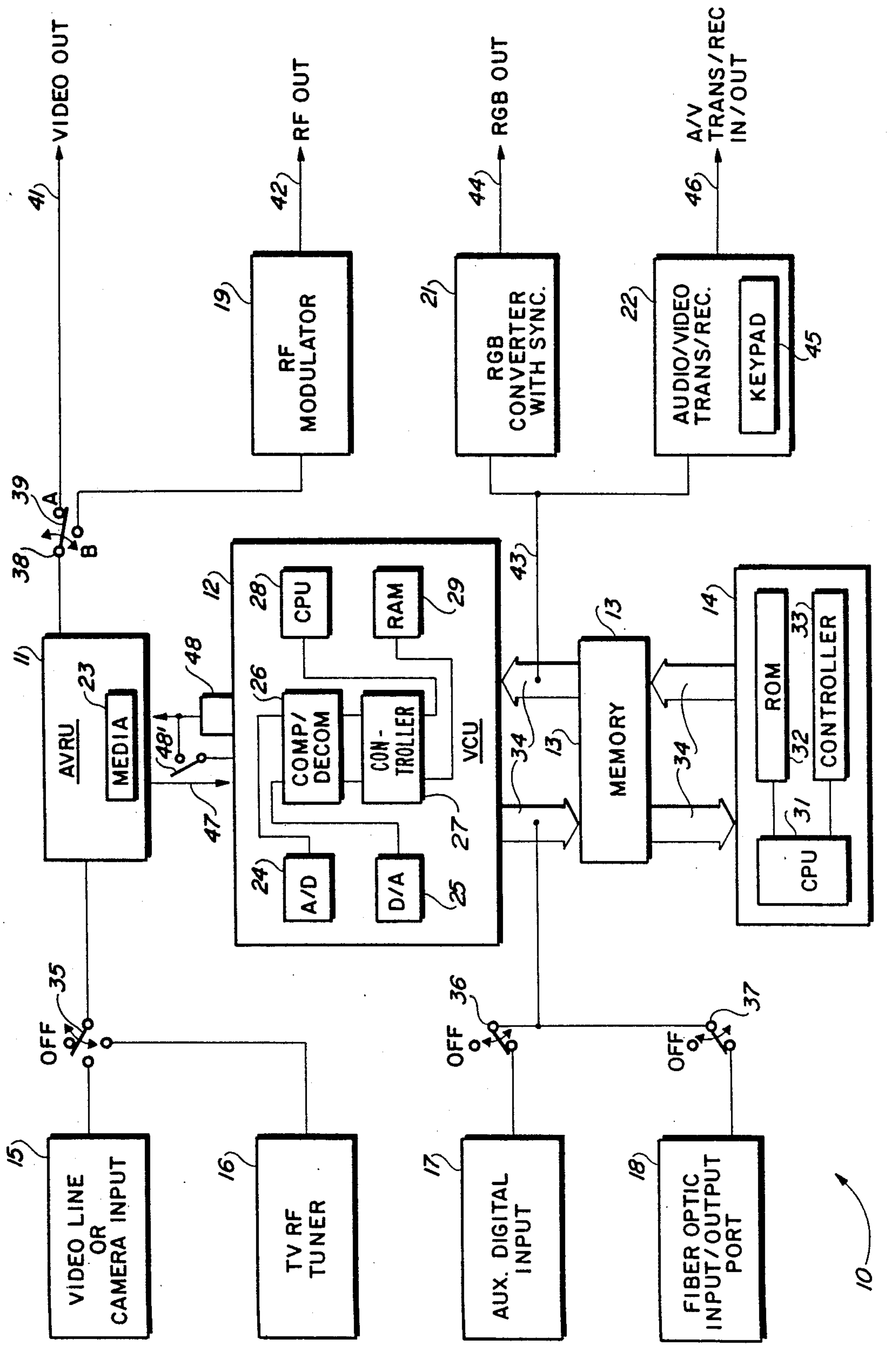
FIG. 1A

FIG. 1



VIDEO OUT
RF OUT
RGB OUT
A/V TRANS/REC IN/OUT
RF MODULATOR
RGB CONVERTER WITH SYNC.
AUDIO/VIDEO TRANS/REC.
KEYPAD
AVRU
MEDIA
VCU
CPU
RAM
COMP/DECOM
CON-TROLLER
A/D
D/A
MEMORY
ROM
CONTROLLER
CPU
OFF
A
B
VIDEO LINE OR CAMERA INPUT
TV RF TUNER
AUX. DIGITAL INPUT
FIBER OPTIC INPUT/OUTPUT PORT
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11
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14
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23
24
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26
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48'

FIG. 2

AUDIO/VIDEO TRANSCEIVER APPARATUS INCLUDING COMPRESSION MEANS

BACKGROUND OF THE INVENTION

The video cassette recorder (VCR) has added significantly to the usefulness of the home television set. Important or exceptionally good programs may be recorded to be viewed again. Programs appearing at times that are inconvenient for viewing may be recorded for playback at a better time. Recorded movies or other materials, educational or entertaining, may be rented or borrowed for viewing at home. (As used in the remainder of this specification, the term "program" encompasses movies and other types of video materials, whether broadcast from a TV station or another source.)

The typical VCR has its own tuner-receiver and a video-recorder. It can receive and record a program from one channel while the television set is being employed to view a program on another channel. Programs are recorded on magnetic tape. The tape is then played back and viewed on the television set. Features commonly included in the VCR are capabilities for advancing the tape forward or backward at a high speed, stopping motion at any frame to hold the image, or simply playing back the recording at normal speed.

Desirable features that are not normally available in a VCR are capabilities for copying recorded programs from one tape or alternative storage medium to a similar or dissimilar storage medium, editing recorded programs and high speed recording. Another desirable but currently unavailable feature is the capability for high speed, high quality transmission and reception by optical fiber using the VCR.

DESCRIPTION OF THE PRIOR ART

U.S. Pat. No. 4,768,110 incorporated herein by reference, describes a VCR having two tape decks included therein. The purpose for the inclusion of two decks rather than the usual single tape deck is to permit the simultaneous viewing of a live RF-modulated TV signal or prerecorded material while recording another live RF-modulated TV signal and to also allow the copying of material from a first magnetic cassette tape onto a second magnetic cassette tape without the use of a second VCR. Viewing of the recorded material during the copying process is also possible in this arrangement. A major disadvantage is that the incorporation of the second tape deck is expensive and limited to magnetic tape, and furthermore, this prior art does not allow for the transmission or reception of recorded material over optical fibers or the high speed reception or transmission of audio/video material in a digital format. An additional disadvantge is the inability for random access editing of the audio/video signal. Furthermore, the additional mechanical structure adds significantly to the overall dimension of the equipment and increases the prospects of mechanical failures.

SUMMARY OF THE INVENTION

In accordance with the invention, an improved audio/video recorder is provided with added features and functions which significantly enhance its usefulness and functionality.

It is, therefore, an object of the present invention to provide an improved audio/video recorder for use in conjunction with an ordinary home television set.

Another object of the invention is to provide in such an improved audio/video recorder a capability for transferring a previously recorded program from one magnetic tape or other storage medium to another.

A further object of the invention is to provide such a capability for transferring a recorded audio/video program without resort to the use of two magnetic tape decks, this being a cumbersome, limited, and expensive approach already proposed in the prior art.

A still further object of the invention is to provide an effective and efficient means for intermediate storage of the audio/video program in digital memory as a means for achieving the transfer of the audio/video program from one tape or storage medium to another.

A still further object of the invention is to provide in such an improved audio/video recorder a capability for accepting various forms of analog or digital audio and video input signals and for converting the analog input signals to digital form when appropriate.

A still further object of the invention is to provide in such an improved audio/video recorder a capability for editing the video input signals without the necessity of using multiple cassettes or recording media.

A still further object of the invention is to provide an improved audio/video recorder for connection to various signal sources including a TV RF tuner, video camera, video line input, and direct audio/video digital input from sources as diverse as a fiber optic input line or a computer.

A still further object of the invention is to provide an improved audio/video recorder having a capability for mixing live audio/video programs with either analog or digital audio/video input signals from another source.

A still further object of the invention is to provide an improved audio/video recorder for simultaneously playing, viewing, recording and/or mixing digital and analog audio/video programs from different digital and analog audio/video sources or storage media.

A still further object of the invention is to provide an improved audio/video recorder which maximizes a given storage capacity, through the use of a data compression technique.

A still further object of the invention is to provide an audio/video recorder utilizing a data compression technique for efficient storage, transmission, and reception of a digitized audio/video program over telephone lines or by other external digital means such as satellite transmission or reception.

A still further object of the invention is to provide in such an improved audio/video recorder a capability for delivering output signals in different forms or formats including a standard RF modulated output signal for viewing on a television set, a digital output signal for viewing on a high-resolution monitor, and audio output signals for a speaker system.

A still further object of this invention is to provide an improved audio/video recorder which provides for random access to any given segment of a self-stored audio/video program so that the desired segment may be accessed and viewed without the time-consuming delays normally involved in fast-forward or fast-reverse searching procedures employed in present state-of-the-art VCR's.

A still further object of the invention is to provide an improved audio/video recorder which provides conve-

nience in the editing of stored data by virtue of its random access memory capability.

A still further object of the invention is to provide an improved audio-video recorder which has the potential for enhanced audio and video quality by virtue of its capability for digital audio/video output and digital filtering techniques.

Further objects and advantages of the invention will become apparent as the following description proceeds, and the features of novelty which characterize the invention will be pointed out with particularity in the claims annexed to and forming a part of this specification.

BRIEF DESCRIPTION OF THE DRAWING

The present invention may be more readily described with reference to the accompanying drawing, in which:

FIG. 1 is a perspective view of the housing of the audio/video recorder disclosed and embodying the invention;

FIG. 1A is an enlarged view of the circled area of FIG. 1; and

FIG. 2 is a functional block diagram of the audio/video recorder of FIG. 1.

DESCRIPTION OF THE PREFERRED EMBODIMENT

Referring to the drawing by characters of reference, FIGS. 1 and 2 illustrate an improved audio/video recorder transmitter-editor 10 (the "VCR-ET") comprising an audio/video recording unit (AVRU) 11, a video control unit (VCU) 12, memory 13, digital control unit (DCU) 14, video line or camera input line 15, TV RF tuner 16, auxiliary digital input port 17, fiber optic input/output port 18, RF modulator 19, RGB converter with syncronizer 21, and an audio/video transmitter/receiver 22 with keypad 45, all in a common housing.

The audio/video recording unit AVRU 11 may be a video cassette recorder similar to a conventional VCR in which the storage media 23 is a magnetic tape. Alternatively AVRU 11 may operate with other types of storage media including, but not limited to, other magnetic tape formats. AVRU 11 has all the functions of the typical VCR including record, play, rewind, slow motion, fast-forward and single frame hold.

An alternate form of storage media for use in AVRU 11 is the CD-ROM, which is a disk using a derivative of glass or plastic in conjunction with an aluminum or other metallic coating. Audio and video signals are stored in the form of irregularities in the aluminum coated surface and are read using a low power laser. In this case, the user would not be able to store or write into the CD-ROM but would be able to play discs that have been recorded and distributed commercially. The storage of video and audio signals on the CD-ROM is in digital form which is readily accommodated by the video recorder of this invention.

Instead of using a CD-ROM, VCR-ET 10 can achieve both record and play capabilities by using optical discs as media 23. Such optical discs are similar to a CD-ROM and use a variable power laser to read from or write on the disc.

A first type of optical disc may comprise a WORM (Write Once Read Many) optical disc. This device has the unique capability of writing on the disc permanently. A laser is used to burn a pit in the media or to change the magnetic or optical properties of the media. A lower-powered laser is then used to read the data from the disc. Data, in this case, is permanently recorded; it may neither be erased nor written over. A further description of this technology can be found in the November 1988 issue of *The Electronic System Design* magazine (ESD) page 55–56.

A second and preferred type of optical disc to be used in AVRU 11 is an erasable optical disc. This disc has full read/write/erase capabilities. With this disc, AVRU 11 has the same record/playback capabilities as a conventional VCR. As an example, erasable optical discs are used in Steven Jobs' "Next" machine as described in *Infoworld,* Volume 10, issue 42, pages 51 and 93, Oct. 17, 1988. In addition, the digital format and the random access capabilities of the erasable disc (and of the CD-ROM and WORM) provide additional benefits as will be discussed in a later part of this specification.

A key element of VCR-ET 10, which is responsible for its improved functionality, is the video control unit or VCU 12. The VCU comprises an analog to digital converter (ADC) 24, a digital to analog converter (DAC) 25, a compressor/decompressor 26, a controller 27, a central processing unit (CPU) 28 and a random access memory (RAM) 29. VCU 12, using these elements, accomplishes the digitization and compression of analog signals as well as the reverse process in which the compressed digital signals are decompressed and converted back to analog signals.

As a first step in the processing of the composite video signals within VCU 12, the sync signals are decoded to isolate signals for each picture frame for processing.

The video signals defining each frame may then be converted to a red analog signal, a green analog signal, and a blue analog signal in a conventional manner. The red, green and blue analog signals are then converted to digital form by the analog to digital converter (ADC) 24. The frame is divided into a set of closely positioned rows and columns of picture elements or "pixels." Each pixel has a color defined by a set of three digital values defining strength of the primary color components, red, green and blue (RGB) respectively. In one embodiment, each frame is divided into an array of 300 by 300 pixels, with the color and luminance of each pixel being defined by a seven bit word for the red component, a seven bit word for the blue component, and a seven bit word for the green component. These words are generated by ADC 24.

If each frame includes 90,000 pixels (300×300), and each pixel is defined by 21 bits (7 bits per primary color), the digital representation of a single video frame utilizes a sizable block of digital information (i.e., 1.89 megabits/frame) which must be processed very rapidly. (Approximately 30 frames/second are received from AVRU 11.) Fortunately the analog to digital conversion of these signals may be accomplished at the desired speed using commercially available analog to digital converter integrated circuits. The analog to digital converter 24 (ADC) is a high-speed, high-accuracy, A to D "flash" converter available as a single IC (integrated cirucit). Several different types of such A/D converters are available from Burr-Brown, one of which is the ADC 600.

Compression of the digital data defining a video frame and the reverse process (decompression) are accomplished by compressor/decompressor 26. Various algorithms may be employed in the compression process which enable the representation of a series of numbers by a reduced number of digits. As an example,

compression algorithms like CCITT Group IV may be used. Existing compression algorithms, like CCITT Group IV, are available on a single integrated circuit. One example of an appropriate compression/decompression circuit on a single integrated circuit is the AMD (Advanced Micro Devices) 7971. The 7971 is described in the Am7971A data sheet, published by AMD in 1988, and incorporated herein by reference.

In one optional embodiment, to further reduce the amount of memory required to store a program the compression algorithm can simply record data corresponding to only those pixels which change color from one frame to the next. This results in considerable memory space savings, since not all pixels change color each frame. Basing calculation upon 10% of the pixels changing from one frame to the next, it is estimated that memory requirements using this technique are cut by about 90%. It is also estimated that on the average, the CCITT Group IV algorithm can cut memory requirements by another 95%. Thus, if no data compression technique is used, it would take approximately 51.03 gigabytes to store a 2 hour movie, but using the above compression techniques, it is estimated that memory **13** will require only 250 megabytes.

Controller **27** handles timing and aids in the communication between the different elements of VCU **12**, and between VCU **12**, AVRU **11** and memory **13**.

In one embodiment, the audio portion of the program is periodically sampled and digitized by digital to analog conversion In one embodiment, this is done at a sample rate of 88,000/second, one byte per sample, to yield CD quality sound. The sampling rate could be dropped to reduce memory requirements. Also, the audio data can be compressed with conventional algorithms, e.g., a Fibonacci delta compression algorithm.

The process of converting either from analog to digital or from digital to analog requires memory for intermediate storage. Random Access Memory(RAM) **29** serve in this capacity. For this purpose either a DRAM (Dynamic RAM) or a SRAM (static RAM) may be employed. An example of a DRAM is the TI (Texas Instruments) TMX4C1024; an example of a SRAM is the INMOS IMS-1203. RAM **29** should have sufficient capacity to store at least two full uncompressed frames (e.g., about 472 KB).

The CPU (Central Processing Unit) **28** is a microprocessor which controls the digitization process of VCU **12**. CPU **28** works with controller **27** to control and communicate with the other elements of the VCU. There are numerous commercially available microprocessors that are appropriate for this application. The Intel 80286, Intel 80386, Motorola 68020, and Motorola 68030 are examples. A more complete description of the microprocessors can be found in the Oct. 27, 1988 issue of *Electronic Design News* (EDN), pages 231 and 242, or in the applicable data sheets.

Controller **27**, CPU **28** and RAM **29** serve in the same manner during the reverse processes, i.e., decompression and digital to analog conversion. Decompression is first accomplished in compressor/decompressor **26**. The decompressed digital signal is then converted to an analog signal by digital to analog converter (DAC) **24** (assuming its destination requires an analog form). In the course of converting the decompressed signals from the VCU **12** for use by the AVRU **11** the signals are synchronized by the time base generator (TBG) or corrector **48**. TBG **48** can be by passed by a shunt switch **48'** for the purpose of transmitting either compressed or decompressed signals from VCU **12** directly to the AVRU **11** in an uncorrected time based mode.

DAC **25** provides the inverse of the function performed by A/D converter **24**. DAC **25** is a high-speed, high accuracy digital to analog converter. An example of such a converter is the Burr-Brown DAC60 digital to analog converter.

Different types of memory technologies are adaptable for use in memory **13**. As mentioned earlier, DRAM and SRAM semiconductor memories are commonly used for applications of this type and are readily available.

One type of random access memory is CMOS (Complimentary Metal Oxide Semiconductor). The CMOS memory has the advantage of a relatively low power requirement and is readily adaptable for use of battery backup for semi-permanent data storage. Another type of memory is the above mentioned optical disc memories.

Emerging memory technologies may also prove advantageous with capabilities for mass data storage in even smaller physical dimensions.

Digital Control Unit (DCU) **14** comprises a CPU (Central Processor Unit) **31**, a ROM (Read Only Memory) **32** and a controller **32**. DCU **14** is responsible for all of the digital editing processes. Through the use of DCU **14**, video segments may be edited and rearranged. Thus, one may use DCU **14** to rearrange the scenes in a movie, alter the movie sound track, etc.

In addition, a program may be edited, one frame at a time, by changing the contrast, brightness, sharpness, colors, etc. (Alteration of the contrast, brightness, sharpness and colors can be automated as well.) Images could be rotated, scaled (i.e., made larger or smaller), etc. In addition, pixel by pixel editing can be accomplished by DCU **14**, e.g., in a manner similar to the PC paint program. Similar editing features can be incorporated for the audio portion of each program. In one embodiment, a display such as a flat panel video display (not shown) is built into the VCR-ET. A user interface control panel of DCU **14** allows a user to select a desired frame number from a menu on the display The VCR-ET then displays a strip of frames (including several frames before and after the selected frame). The user can delete frames in a strip, select a point where other frames are to be inserted into the program, or enhance different frames. A light pen or mouse ca be used to select individual frames in a strip.

Instead of incorporating a flat display into VCR-ET **10**, in another embodiment, a television coupled to output lead **42** of RF modulator **19** can be used during editing.

CPU **31** is a microprocessor of the type described in connection with the CPU **28** of VCU **12**. Controller **33** is a integrated circuit which handles the timing and aids in communication between DCU **14** and memory **13**. ROM **32** holds the necessary step-by-step editing programs which are installed at the factory. A currently available example of a suitable ROM for this application is the Texas Instruments part TMS47256. CPU **31** and controller **33** together control the editing process as they execute the programs stored in ROM **32**.

The VCU **12**, memory **13** and DCU **14** communicate with each other via a high speed data bus **34**. The high speed data bus is required in order to meet bandwidth requirements. Examples of suitable data bus devices are Motorola's VME bus, Intel's Multibus and the Optobuss (U.S. Pat. No. 4,732,446).

A video line or camera input line **15** is provided to enable VCR-ET **10** to receive an input signal from a source such as a television camera, a conventional VCR, a television tuner, or another VCR, etc. The signals received at input line **15** are typically carried by a coaxial cable and are in the form of a standard television composite signal. As used throughout this specification, the words "standard television composite signal" or its acronym STCS shall be read to mean any one of the following: NTSC, PAL, SECAM, HDTV, or any American or European broadcast signal standards. An NTSC composite signal is defined as the analog signal that carries the chrominance (color), luminance (brightness), synchronization (timing) and audio signals that make up the video signals received an displayed by television and video cassette recorders. These four components are combined into one signal by modulating the components in different ways. (Amplitude modulation and phase modulation are examples.) The standard video line signal is such a composite signal and may be received at input line **15** from one of the above-mentioned sources.

TV RF tuner input port **16** also supplies a composite signal as described in regard to video input line **15**. The difference is that this signal is received from an antenna or cable TV coaxial cable. To receive such a signal, tuner **16** is capable of being set or tuned to receive the desired carrier frequency or television channel.

Selector switch **35** is provided to select either video input line **15** or TV RF tuner **16** as an input signal source to AVRU **11**.

Auxiliary digital input port **17** is employed to receive any acceptable digital signal such as computer-generated video signal or as may be supplied by another VCR-ET. This signal, for example, may be an RGB video signal such as that delivered to computer monitors, or it may be a digitized audio signal. (As mentioned above, an RGB signal is a signal which communicates the strength of the red, green and blue color components for the pixels that make up each video frame.) Switch **36** selects whether the digital video/audio input signal is chosen from auxiliary digital input port **17**. Switch **36** supplies the selected signal to high speed data bus **34** which carries the signals in digital form.

Fiber optic port **18** incorporates a fiber optic transceiver/receiver. Port **18** has a capability for transforming fiber optic (light) signals to electrical signals or for transforming electrical signals to fiber optic signals. Port **18** thus provides a capability for two-way communication between high speed data bus **34** and a fiber optic signal line. The incorporation of fiber optic port **18** in the VCR-ET provides a capability for receiving audio/video signals from or delivering audio/video signals to the fiber optic line such as a fiber optic telephone line. The fiber optic line carries digital signals in the form of light waves over great distances with a high degree of accuracy and reliability and at a high speed (e.g., about 200 megabytes/second). The VCR-ET can receive a video program at an accelerated rate via fiber optic port **18**, e.g., from a variety of sources. For example—a video program may be communicated at an accelerated rate from the first VCR-ET to a second VCR-ET in less time than it would take to view the program. Thus, it is not necessary to access the optical fiber for long periods of time to transmit a long video program.

It is also envisioned that in the future, a video library may be established which downloads video programs at an accelerated rate via optical fibers to a subscriber's VCR-ET.

Switch **37** is provided to select connection to the fiber optic input/output port **18**. An OFF or open position is provided. The selected signal is delivered to or supplied from high speed data bus **34**.

Analog output signals from AVRU **11** are delivered to the common terminal **38** of a selector switch **39**. When set to position A, switch **39** delivers the output signal of AVRU **11** directly to a video output line **41** as a standard STCS composite signal; when set to position B switch **39** delivers the output of VRU **11** to the input of RF modulator **19**. Modulator **19** converts the video signal to an RF-modulated composite signal for delivery to such devices as televisions and conventional VCR's. These types of devices play back the video program on a particular frequency channel (such as channel 4) on the television. Delivery to the television or VCR is via RF output line **42**.

Digital output signals from VCR-ET **10** may be dispatched from high speed data bus **34** via line **43** to input leads of RGB converter **21** and audio-video transmitter/receiver **22**.

RGB converter **21** converts the STCS signal into an RGB signal as required by computer monitors and similar display devices. The converted signal is received by a display device connected to RGB converter output line **44**.

VCR-ET **10** includes audio/video transmitter/-receiver **22** which is typically a modem. Advantageously, the modem may be used to communicate an audio/video program over conventional phone lines in a manner similar to that described above with respect to optical fibers. The term modem is derived directly from its functionality as a modulator-demodulator which allows transfer of the audio/video signal over the standard telephone line. Modems are commonly available for computers and are currently available in the form of a single integrated circuit. As an example, Sierra Semiconductor offers a 2400 baud single chip modem under its part number SC111006. Representative manufacturers of these single modem IC's can be found in the Apr. 14, 1988 issue of Engineering Design News (EDN), pages 124–125. Some of these single modem IC's have the added capability of generating the tones for dialing a phone number. The destination phone number ma be entered by means of an optional keyboard/keypad **45** incorporated in the video recorder **10** of the invention. Output port **46** of transmitter/receiver **22** connects directly to the telephone line. (It is noted that the band width of a conventional phone line is at present much narrower than the signal band width of an optical fiber, and thus the data transmission rate on telephone lines is much slower than the transmission rate for an optical fiber. Accordingly, the time required to communicate a video program over a conventional phone line may exceed the time it takes to view the program.)

The application and utilization of the VCR-ET ma include a number of forms or operating modes.

In its first and simplest operating mode, AVRU **11** may be operated in the manner of a conventional VCR with signals from an antenna being received by tuner **16** and recorded directly on media **23** in analog form. At the same time the received program may be viewed on the television screen with the television connected at video output terminal **41**. An optional signal source for this type of operation is the video line or camera input line **15** selectable by switch **35**.

In a second operating mode a program stored on media **23** of AVRU **11** may be played back and viewed on the connected television set.

When it is desired to copy a program from one recording media to another, the recording media holding the desired program is installed in the AVRU. The recording media is then played back with optional viewing on a connected television set o other TV monitor or listening through speakers (as appropriate). As the recording media is played back, the analog signals from the recording media (video and/or audio) are dispatched to VCU **12** via connection **47**. The analog signals are converted to digital signals by ADC **24**, compressed by compressor/decompressor **26** and the compressed digital signals are stored in memory **13**. The foregoing operations are accomplished under the control of controller **27** and CPU **28**. RAM **29** is used for interim data storage during this process. Once the complete video/audio program has been stored in memory **13**, the recording media from which the stored program has just been read is replaced by blank recording media upon which the stored program is to be copied. CPU **28** in cooperation with controller **27** and RAM **29** then executes the decompression and digital to analog conversion of the program stored in memory **13**, decompression taking place in compressor/decompressor **26**, and digital to analog conversion being accomplished by DAC **25**. The resulting analog program is stored on the blank recording media which constitutes media **23** of AVRU **11**.

During the foregoing copying procedures, DCU **14** may be utilized for editing operations. As the program is being read from the first or original recording media, it is simultaneously viewed on the TV screen, or listened to by means of an audio monitor, converted to digital signals, compressed and stored in memory **13**. Once the digital audio/video program is stored in memory **13**, editing is accomplished by the user through control of DCU **14**, by means of a control panel (not shown) coupled to DCU **14**. If desired, additional audio/video signals may be simultaneously entered into memory **13** and added to those received from VCU **12**. The additional signals ma be introduced from auxiliary digital input port **17** or from fiber optic input/output port **18** and may comprise video captions for superimposed position upon the stored video images, or they may be audio commentaries to be added to silent video presentations. In addition, as mentioned above, the order in which various segments appear in the video programs ma be altered. Certain undesired segments, such as TV commercials, may be removed. This editing operation is accomplished under the control of DCU **14**.

In still another operating mode a program stored in media **23** of AVRU **11** or being received by AVRU **11** from input line **15** (as from a video camera) may be digitized and compressed by VCU **12** and routed via bus **34**, to memory **13**. The data from memory **13** is then routed to line **43**, transmitter/receiver **22** and to a telephone line. At the other end of the telephone line the signals received are processed by another VCR-ET. As indicated above, conventional nonoptical telephone lines do not typically support high data transmission rates at the present time. Accordingly, even compressed data may require more time to transmit over conventional phone lines than it would take to view the actual video program.

Once received in the second VCR-ET's memory **13**, the digitized program ca then either be viewed directly from memory or transferred to storage medium **23**, either in its entirety or in random segments, based on user preference.

In the case of video camera input at input **15** the transmitted signals may comprise a live transmission. Alternatively the transmitted program may be derived from a program stored on media **23** of AVRU **11**. In this case the stored analog program is again decoded, digitized, compressed and transmitted via bus **34** to memory **13**. The data in memory **13** is then communicated via line **43** and transmitter/receiver **22** to telephone lines.

It follows, of course, that digitized video and audio signals from the remote VCR-ET at the far end of the telephone line may be received at line **46**, entered into memory **13** via transmitter/receiver **22**, converted to analog signals by VCU **12**, and recorded on media **23** and then viewed, if desired, on a television set connected at output **41**.

As mentioned earlier, when any of the foregoing operations entail the processing of unmodulated video signals, such signals must first be processed by RF modulator **19** before they can be accepted by devices such as a conventional VCR or television set; when the monitoring means is a computer monitor or a similar display device the signals are processed by RGB converter **21**.

All of the foregoing operations are performed with enhanced quality and efficiency by virtue of the digital, rather than analog, storage and transmission modes and the compressed data storage mechanism, with additional advantages of improved cost and reliability afforded in the case of tape to tape (or other media to media) program transfers by virtue of the requirement for only a single tape deck or other storage device.

All of the foregoing operations, to the extent they relate to the editing, playback, reception and/or transmission of video signals are also analogous to the VCR-ET's capabilities with regard to analog or digital signals containing only audio material.

An improved audio/video recorder with significantly expanded functional capabilities is thus provided in accordance with the stated objects of the invention and although but a single embodiment of the invention has been illustrated and described, it will be apparent to those skilled in the art that various changes and modifications may be made therein without departing from the spirit of the invention or from the scope of the appended claim. For example, the VCR-ET can be constructed so as to be portable. Thus, it could be carried to a location along with a video camera where it is desired to record a program, and then taken to another location where it is used to edit the program. Other modifications will be apparent to those skilled in the art in light of the present specification.

What is claimed is:

1. An audio/video transceiver apparatus comprising:

input means for receiving audio/visual source information;

compression means, coupled to said input means, for compressing said audio/video source information into a time compressed representation thereof having an associated time period that is shorter than a time period associated with a real time representation of said audio/video source information;

random access storage means, coupled to said compression means, for storing the time compressed

representation of said audio/video source information; and

output means, coupled to said random access storage means, for receiving the time compressed audio/video source information stored in said random access storage means for transmission away from said audio/video transceiver apparatus.

2. An audio/video transceiver apparatus as in claim 1 further comprising editing means, coupled to said random access storage means, for editing the time compressed representation of said audio/video source information stored in said random access storage means and for restoring the edited time compressed representation of said audio/video source information in said random access storage means; and wherein said output means is operative for receiving the edited time compressed representation of said audio/video source information stored in said random access storage means for transmission away from said audio/video transceiver apparatus.

3. An audio/video transceiver apparatus as in claim 2 further comprising monitor means for enabling the user to selectively identify the time compressed representation of said audio/video source information stored in said random access storage means during editing.

4. An audio/video transceiver apparatus as in claim 1 wherein said output means comprises a fiber optic output port for coupling said audio/video transceiver apparatus to a fiber optic transmission line.

5. An audio/video transceiver apparatus as in claim 1 wherein said output means comprises a modem for coupling said audio/video transceiver apparatus to a telephone transmission line.

6. An audio/video transceiver apparatus as in claim wherein said random access storage means comprises an optical disc.

7. An audio/video transceiver apparatus as in claim 1 wherein said random access storage means comprises a semiconductor memory.

8. An audio/video transceiver apparatus as in claim 1 wherein:

said audio/video source information comprises analog audio/video source information;

said audio/video transceiver apparatus further comprises analog to digital converter means for converting said analog audio/video source information to corresponding digital audio/video source information;

said compression means is operative for compressing said corresponding digital audio/video source information into a digital time compressed representation thereof having an associated time period that is shorter than a time period associated with a real time representation of said digital audio/video source information; and

said random access storage means is operative for storing said digital time compressed representation of said corresponding digital audio/video source information.

9. An audio/video transceiver apparatus as in claim 1 wherein:

said audio/video source information comprises digital audio/video source information;

said compression means is operative for compressing said digital audio/video source information into a digital time compressed representation thereof having an associated time period that is shorter than a time period associated with a real time representation of said digital audio/video source information; and

said random access storage means is operative for storing said digital time compressed representation of said digital audio/video source information;

10. An audio/video transceiver apparatus as in claim 8 wherein said input means is coupled to an external television camera and said analog audio/video source information comprises information received from said external television camera.

11. An audio/video transceiver apparatus as in claim 8 wherein said input means is coupled to an external analog video tape recorder and said analog audio/video source information comprises information received from said external analog video tape recorder.

12. An audio/video transceiver apparatus as in claim 8 wherein said input means is coupled to an external television RF tuner and said analog audio/video source information comprises information received from said external television RF tuner.

13. An audio/video transceiver apparatus as in claim 8 wherein said input means comprises television RF tuner means coupled to an external television antenna and said analog audio/video source information comprises information transmitted by a remotely located television transmitter.

14. An audio/video transceiver apparatus as in claim 8 wherein said input means comprises television RF tuner means coupled to an external cable television system and said analog audio/video source information comprises information received from said external cable television system.

15. An audio/video transceiver apparats is in claim 9 wherein said input means is coupled to an external computer and said digital audio/video source information comprises computer-generated audio/video information.

16. An audio/video transevier apparatus as in claim 9 wherein said input means comprises a fiber optic input port coupled to a fiber optic transmission line and said digital audio/video source information comprises information received over said fiber optic transmission line.

17. An audio/video transceiver apparatus comprising:

input means for receiving audio/video source information as a time compressed representation thereof, said time compressed representation of said audio/video source information being received over an associated burst time period that is shorter than a real time period associated with said audio/video source information;

random access storage means, coupled to said input means, for storing the time compressed representation of said audio/video source information received by said input means; and

output means, coupled to said random access storage means, for receiving the time compressed representation of said audio/video source information stored in said random access storage means for transmission away from said audio/video transceiver apparatus.

18. An audio/video transceiver apparatus as in claim 17 wherein:

said input means comprises a fiber optic input port;

said input means is coupled, via a fiber optic transmission line, to a video library, said video library storing a multiplicity of items of audio/video source information in said time compressed representation

for selective retrieval, in said associated burst time period over said fiber optic transmission line, by the user.

19. An audio/video transceiver apparatus as in claim **17** in combination with a video library, coupled via a communication link with said audio/video transceiver apparatus, said video library storing a multiplicity of items of audio/video source information in said time compressed representation for selective retrieval, in said associated burst time period over said communication link.

20. An audio/video transceiver apparatus as in claim **1** further comprising:

decompression means, coupled to said random access storage means, for selectively decompressing said time compressed representation of said audio/video source information stored in said random access storage means; and

editing means, coupled to said random access storage means and decompression means, for editing said selectively decompressed time compressed representation of said audio/video source information, and for storing said edited selectively decompressed time compressed representation of said audio/video source information in said random access storage means.

21. An audio/video transceiver apparatus as in claim **1** further comprising:

decompression means, coupled to said random access storage means, for selectively decompressing said time compressed representation of said audio/video source information stored in said random access storage means; and

editing means, coupled to said random access storage means an decompression means, for editing said selectively decompressed time compressed representation of said audio/video source information;

wherein said compression means is operative for recompressing the edited selectively decompressed time compressed representation of said audio/video source information; and

wherein said random access storage means is operative for storing the recompressed selectively decompressed time compressed representation of said audio/video source information.

22. An audio/video transceiver apparatus as in claim **1** further comprising:

decompression means, coupled to said random access storage means, for selectively decompressing the time compressed representation of said audio/video source information stored in said random access storage means; and

monitor means for enabling the user to view the selectively decompressed time compressed representation of said audio/video source information.

23. An audio/video transceiver apparatus as in claim **8** further comprising:

decompression means, coupled to said random access storage means, for selectively decompressing the digital time compressed representation of said corresponding digital audio/video source information stored in said random access storage means; and

editing means, coupled to said random access storage means and decompression means, for editing the decompressed digital time compressed representation of said corresponding digital audio/video source information and for then storing the edited decompressed digital time compressed representation of said corresponding digital audio/video source information in said random access storage means.

24. An audio/video transceiver apparatus as in claim **23** further comprising monitor means for enabling the user to selectively view the decompressed digital time compressed representation of said corresponding digital audio/video source information during editing.

25. An audio/video transceiver apparatus as in claim **8** further comprising:

decompression means, coupled to said random access storage means, for selectively decompressing the digital time compressed representation of said corresponding digital audio/video source information stored in said random access storage means; and

monitor means, coupled to said decompression means, for enabling the user to selectively view the decompressed digital time compressed representation of said corresponding digital audio/video source information.

26. An audio/video transceiver apparatus as in claim **9** further comprising:

decompression means, coupled to said random access storage means, for selectively decompressing the digital time compressed representation of said digital audio/video source information stored in said random access memory means; and

editing means, coupled to said random access storage means and decompression means, for editing the decompressed digital time compressed representation of said digital audio/video source information;

said random access storage means thereafter being operative for storing the edited decompressed digital time compressed representation of said digital audio/video source information in said random access storage means.

27. An audio/video transceiver apparatus as in claim **26** further comprising monitor means for enabling the user to selectively view the decompressed digital time compressed representation of said digital audio/video source information during editing.

28. An audio/video transceiver apparatus as in claim **9** further comprising:

decompression means, coupled to said random access storage means, for selectively decompressing the digital time compressed representation of said digital audio/video source information stored in said random access memory means; and

monitor means, coupled to said decompression means, for enabling the user to selectively view the decompressed digital time compressed representation of said digital audio/video source information.

29. An audio/video transceiver apparatus as in claim **8** further comprising a video tape recorder for providing said analog audio/video source information.

30. An audio/video information transfer network comprising:

a plurality of audio/video transceivers, coupled via one or more communication links, each of said audio/video transceivers comprising;

input means for receiving audio/video source information;

compression means, coupled to said input means, for compressing said audio/video source information into a time compressed representation thereof having an associated burst time period that is shorter than a time period associated with a real time representation of said audio/video source information;

random access storage means, coupled to said compression means, for storing the time compressed representation of said audio/video source information; and

output means, coupled to said random access storage means and to one of said one or more communications links, for receiving the time compressed format representation of said audio/video source information stored in said random access storage means for transmission in said burst time period to another one of said plurality of audio/video transceivers.

31. An audio/video information transfer network as in claim **30** wherein said input means of one of said plurality of audio/video transceivers comprises a fiber optic input port, said output means of another one of said plurality of audio/video transceivers comprises a fiber optic output port, and one of said one or more communications links comprises a fiber optic transmission line coupled between said fiber optic input port and said fiber optic output port.

32. An audio/video information transfer network as in claim **30** wherein said output means of one of said plurality of audio/video transceivers comprises a modem and one of said one or more communications links comprises a telephone transmission line.

33. An audio/video information transfer network as in claim **30** wherein said random access storage means comprises an optical disc memory.

34. An audio/video information transfer network as in claim **30** wherein said random access storage means comprises a semiconductor memory.

35. An audio/video information transfer network as in claim **30** wherein said random access storage means of one of said plurality of audio/video transceivers stores a library comprising a multiplicity of items of audio/video source information in said time compressed representation for selective transmission in said associated burst time period to another on of said audio/video transceivers.

36. An audio/video information transfer network as in claim **30** wherein at least one of said audio/video transceivers further comprises recording means, including a removable recording medium, coupled to said random access storage means, for storing the time compressed representation of said audio/video source information stored in said random access storage means onto said removable recording medium.

37. An audio/video information transfer network as in claim **30** wherein at least one of said audio/video transceivers further comprises:

decompression means, coupled to said random access storage means, for decompressing the time compressed representation of said audio/video source information stored in said random access storage means; and

recording means, including a removable recording medium coupled to said decompression means, for storing the decompressed time compressed format representation of said audio/video source information onto said removable recording medium.

38. An audio/video information transfer network as in claim **36** wherein said recording means comprises a video tape recorder and said removable recording medium comprises magnetic tape.

39. An audio/video information transfer network as in claim **37** wherein said recording means comprises a video tape recorder and said removable recording medium comprises magnetic tape.

40. An audio/video information transfer network as in claim **36** wherein said recording means comprises a write once read many (WORM) optical disc drive and said removable recording medium comprises one or more WORM discs.

41. An audio/video information transfer network as in claim **37** wherein said recording means comprises a write once read many (WORM) optical disc drive and said removable recording medium comprises one or more WORM discs.

42. An audio/video information transfer network as in claim **36** wherein said recording means comprises an erasable optical disc drive and said hard copy storage medium comprises one or more erasable optical discs.

43. An audio/video information transfer network as in claim **37** wherein said recording means comprises an erasable optical disc drive and said hard copy storage medium comprises one or more erasable optical discs.

44. An audio/video transceiver apparatus as in claim **1** further comprising recording means, including a removable recording medium coupled to said random access storage means, for storing the time compressed representation of said audio/video source information stored in said random access storage means onto said removable recording medium.

45. An audio/video transceiver apparatus as in claim **2** further comprising recording means, including a removable recording medium, coupled to said random access storage means, for storing the edited time compressed representation of said audio/video source information stored in said random access storage means onto said removable recording medium.

46. An audio/video transceiver apparats as in claim **45** further comprising monitor means for enabling the user to selectively view the time compressed representation of said audio/video source information stored on said removable recording medium.

47. An audio/video transceiver apparatus as in claim **17** further comprising recording means, including a removable recording medium, coupled to said random access storage means, for storing the time compressed representation of said audio/video source information stored in said random access storage means onto said removable recording medium.

48. An audio/video transceiver apparatus as in claim **2** further comprising recording means, including a removable recording medium, coupled to said random access storage means, for storing the edited decompressed time compressed representation of said audio/video source information stored in said random access storage means.

49. An audio/video transceiver apparatus as in claim **1** further comprising:

decompression means, coupled to said random access storage means, for selectively decompressing the time compressed representation of said audio/video source information stored in said random access storage means; and

recording means, including a removable recording medium, coupled to said decompression means, for storing the selectively decompressed tome compressed representation of said audio/video source information stored in said random access storage means.

50. An audio/video transceiver apparatus as in claim **22** further comprising:

recording means, including a removable recording medium, coupled to said decompression means, for storing the selectively decompressed tome compressed representation of said audio/video source information on said hard copy storage medium; and

wherein said monitor means is operative for enabling the user to view the selectively decompressed time compressed representation of said audio/video source information stored on said removable recording medium.

51. An audio/video transceiver apparatus as in claim **9** further comprising CD-ROM means for providing said digital audio/video source information.

52. An audio/video transceiver apparats as in claim **9** further comprising erasable optical disc means for providing said digital audio/video source information.

53. An audio/video transceiver apparatus as in claim **17** wherein:

said input means comprises television RF tuner means; and

said audio/video source information comprises a time compressed representation thereof transmitted by a remotely located television transmitter.

54. An audio/video transceiver apparatus as in claim **1** further comprising external video tape recorder means, coupled to said output means, for storing the time compressed representation of said audio/video source information stored in said random access storage means onto magnetic tape.

55. An audio/video transceiver apparatus as in claim **2** further comprising external video tape recorder means, coupled to said output means, for storing the edited time compressed representation of said audio/video source information stored in said random access storage means onto magnetic tape.

56. An audio/video transceiver apparatus as in claim **17** further comprising external video tape recorder means, coupled to said output means, for storing the time compressed representation of said audio/video source information stored in said random access storage means onto magnetic tape.

57. An audio/video transceiver apparatus as in claim **52** further comprising external video tape recorder means, coupled to said output means, for storing the edited decompressed time compressed representation of said audio/video source information stored in said random access storage means onto magnetic tape.

58. An audio/video transceiver apparatus as in claim **1** further comprising:

decompression means, coupled to said random access storage means, for selectively decompressing the time compressed representation of said audio/video source information stored in said random access storage means; and

external video tape recorder means, coupled to said output means, for storing the selectively decompressed time compressed representation of said audio/video source information stored in said random access storage means.

59. An audio/video transceiver apparatus as in claim **22** further comprising external video tape recorder means, coupled to said output means, for storing the selectively decompressed time compressed representation of said audio/video source information onto magnetic tape.

60. An audio/video transceiver apparatus comprising:

input means for receiving analog and/or digital audio/video source information;

analog to digital converter means for converting analog audio/video source information received at said input means to corresponding digital audio/video source information;

digital to analog converter means for converting digital audio/video source information received at said input means to corresponding analog audio/video information;

compressor/decompressor means for compressing digital audio/video source information received at said input means or said corresponding digital audio/video source information received from said analog to digital converter means into a time compressed representation of said digital or corresponding digital audio/video source information, said time compressed representation having an associated time period that is shorter than a time period associated with a real time representation of said digital or corresponding digital audio/video source information, said compressor/decompressor means being further operative for decompressing said time compressed representation into a decompressed real time representation of said digital or corresponding digital audio/video source information;

central processing unit means for controlling operation of said compressor/decompressor means;

random access storage means for storing said time compressed representation of said digital or corresponding digital audio/video source information and for storing said decompressed real time representation of said digital or corresponding digital audio/video source information;

controller means for enabling communication between said compressor/decompressor means, said central processing unit means, and said random access memory means; and

output means for receiving said time compressed representation of said digital or corresponding digital audio/video source information stored in said random access storage means for transmission away from said audio/video transceiver apparatus.

61. An audio/video transceiver apparatus as in claim **60** further comprising time bas generator means for supplying timing information for association with said time compressed representation of said digital or corresponding digital audio/video source information.

62. An audio/video transceiver apparatus as in claim **60** further comprising audio/video recording means, including a recording medium, for recording said analog or corresponding analog audio/video source information onto said recording medium.

63. An audio/video transceiver apparatus as in claim **60** further comprising audio/video recording means, including a recording medium, for recording said digital or corresponding digital audio/video source information onto said recording medium.

64. An audio/video transceiver apparatus as in claim **62** wherein said recording medium comprises magnetic tape.

65. An audio/video transceiver apparatus as in claim **63** wherein said recording medium comprises magnetic tape.

66. An audio/video transceiver apparatus as in claim **63** wherein said recording medium comprises a CD-ROM.

67. An audio/video transceiver apparatus as in claim 63 wherein said recording medium comprises a WORM optical disc.

68. An audio/video transceiver apparatus as in claim 63 wherein said recording medium comprises an erasable optical disc.

69. An audio/video transceiver apparatus as in claim 60 further comprising audio/video recording and playback means coupled to said input means for providing said analog and/or digital audio/video source information.

70. An audio/video transceiver apparatus as in claim 60 further comprising high speed bus means coupled to said input means, and wherein said input means comprises auxiliary digital input means for receiving said digital audio/video source information.

71. An audio/video transceiver apparatus as in claim 70 wherein said high speed bus means comprises an optical bus.

72. An audio/video transceiver apparatus as in claim 60 further comprising high speed bus means coupled to said input means, and wherein said input means comprises fiber optic input means for receiving said digital audio/video source information.

73. An audio/video transceiver apparatus as in claim 60 further comprising high speed bus means, and wherein said analog to digital converter means, digital to analog converter means, compressor/decompressor means, central processing unit means, and controller means are coupled to said random access storage means via said high speed bus means.

74. An audio/video transceiver apparatus as in claim 60 further comprising:

digital control unit means, said digital control unit means comprising:

additional central processing unit means;

read-only memory means coupled to said additional central processing unit means for storing microinstructions defining a plurality of selected editing functions; and

additional controller means for enabling communication between said additional central processing unit means and said read-only memory means;

said additional central processing unit means being operative for selectively executing the microinstructions stored in said read-only memory means to perform one or more of said plurality of selected editing functions.

75. An audio/video transceiver apparatus as in claim 74 wherein said digital control unit means is coupled to said random access storage means.

76. An audio/video transceiver apparatus as in claim 73 further comprising RGB converter means for converting information stored in said random access storage means to an RGB format, and wherein said output means comprises RGB output means for receiving RGB format information from said RGB converter means.

77. An audio/video transceiver apparatus as in claim 73 wherein said output means comprises audio/video transmitter/receiver means coupled to said high speed bus for receiving said time compressed representation of said digital or corresponding digital audio/video source information stored in said random access storage means for transmission away from said audio/video transceiver apparatus.

78. An audio/video transceiver apparatus as in claim 77 wherein said audio/video transmitter/receiver means comprises a modem for coupling to a telephone transmission line.

79. An audio/video transceiver apparatus as in claim 77 wherein said audio/video transmitter/receiver means comprises a fiber optic transceiver for coupling to a fiber optic transmission line.

80. An audio/video transceiver apparatus as in claim 1 further comprising editing means, coupled to said random access storage means, for editing said time compressed representation of said audio/video source information and for then storing the edited time compressed representation of said audio/video source information in said random access storage means.

* * * * *

EXHIBIT B

US005164839A

United States Patent [19]

Lang

[11] **Patent Number: 5,164,839**

[45] **Date of Patent: Nov. 17, 1992**

[54] **METHOD FOR HANDLING AUDIO/VIDEO SOURCE INFORMATION**

[75] Inventor: **Richard A. Lang,** Cave Creek, Ariz.

[73] Assignee: **Explore Technology, Inc.,** Scottsdale, Ariz.

[21] Appl. No.: **775,182**

[22] Filed: **Oct. 11, 1991**

Related U.S. Application Data

[60] Division of Ser. No. 347,629, May 5, 1989, Pat. No. 5,057,932, which is a continuation-in-part of Ser. No. 289,776, Dec. 27, 1988, Pat. No. 4,963,995.

[51] **Int. Cl.**^{5} **H04N 5/76**

[52] **U.S. Cl.** **358/335;** 358/133; 360/8; 360/9.1; 360/14.1

[58] **Field of Search** 358/335, 133, 903, 901, 358/310, 134; 360/8, 9.1, 13, 14.1

[56] **References Cited**

U.S. PATENT DOCUMENTS

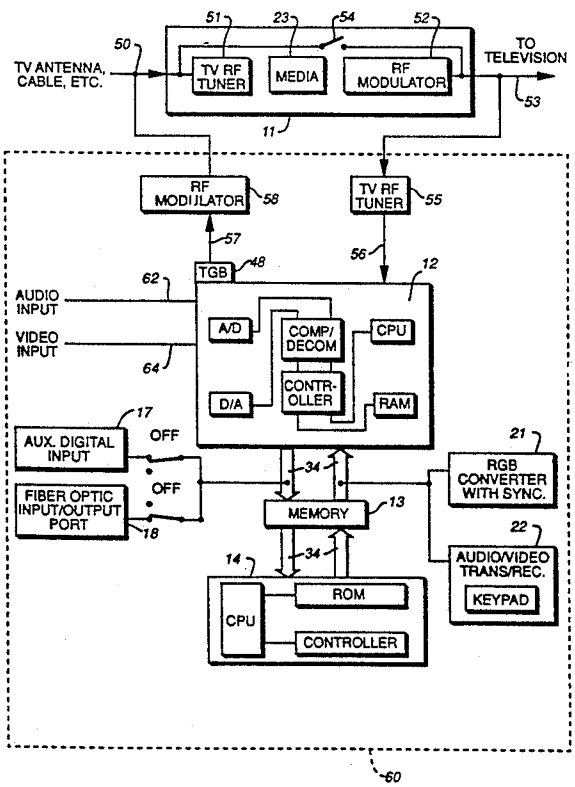
4,179,709	4/1979	Workman	353/133
4,400,717	5/1983	Southworth et al.	358/13
4,446,490	4/1984	Hoshimi et al.	360/32
4,506,387	7/1985	Walter	455/612
4,511,934	3/1985	Ohira et al.	360/55
4,516,156	4/1985	Fabris et al.	358/85
4,563,710	9/1986	Baldwin	360/9.1
4,625,080	12/1986	Scott	379/104
4,654,484	3/1987	Reiffel et al.	379/53
4,698,664	5/1987	Nichols et al.	358/10
4,709,418	8/1987	Fox et al.	455/612
4,724,491	2/1988	Lambert	358/310
4,736,239	7/1988	Sprague et al.	358/21 R
4,743,959	9/1988	Frederiksen	358/11
4,750,034	9/1988	Lem	358/84
4,768,110	8/1988	Dunlap et al.	360/15
4,774,574	9/1988	Daly et al.	358/133
4,821,208	4/1989	Ryan et al.	364/518
4,851,931	7/1989	Parker et al.	360/15
4,868,653	9/1989	Golin et al.	358/133
4,918,523	4/1990	Simon et al.	358/133
5,006,936	4/1991	Hooks, Jr.	360/8

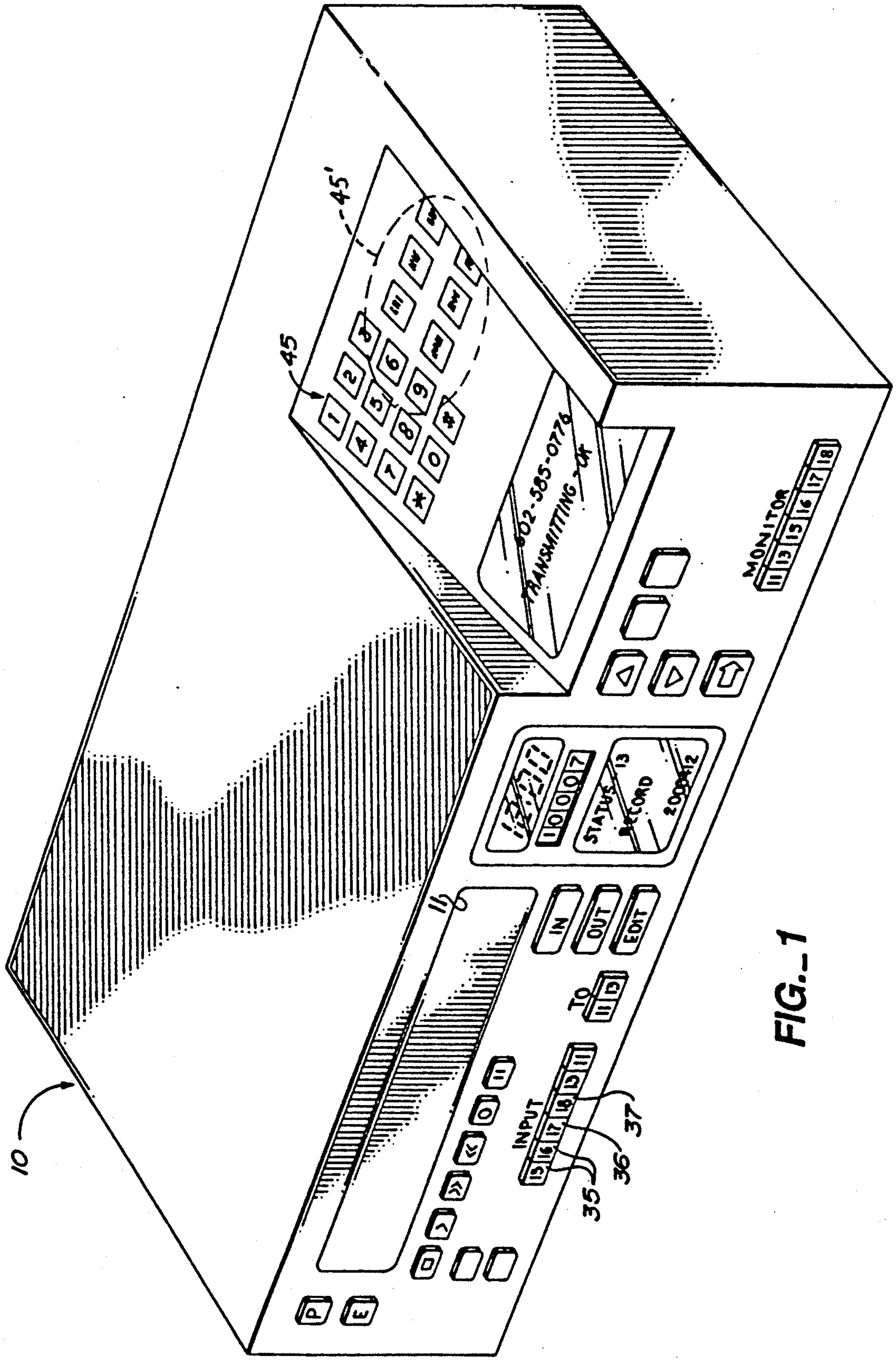
Primary Examiner—Roy N. Envall, Jr.
Assistant Examiner—Huy Nguyen
Attorney, Agent, or Firm—William E. Hein

[57] **ABSTRACT**

An improved video recorder/transceiver with expanded functionality ("VCR-ET") including a capability for storing video and video programs in digital format, editing such programs, transferring such programs onto a hard copy magnetic media, and transmitting such programs to a remote location using a second VCR-ET. The increased functionality is realized through the use of analog to digital conversion, signal compression and intermediate storage in an integrated circuit, random access memory. The recorder/transmitter has capabilities to transmit and receive program information in either a compressed or decompressed format over fiber optic lines, conventional phone lines or microwaves.

77 Claims, 4 Drawing Sheets




10
45
45'
11
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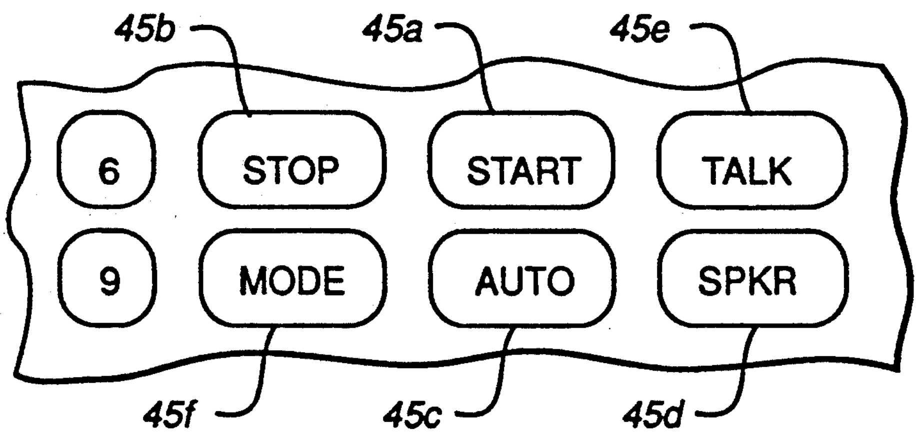
TRANSMITTING - OK
STATUS 13
RECORD
MONITOR
INPUT
IN
OUT
EDIT
TO
35
36
37

FIG._1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START
TALK
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MODE
AUTO
SPKR
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FIG._1A

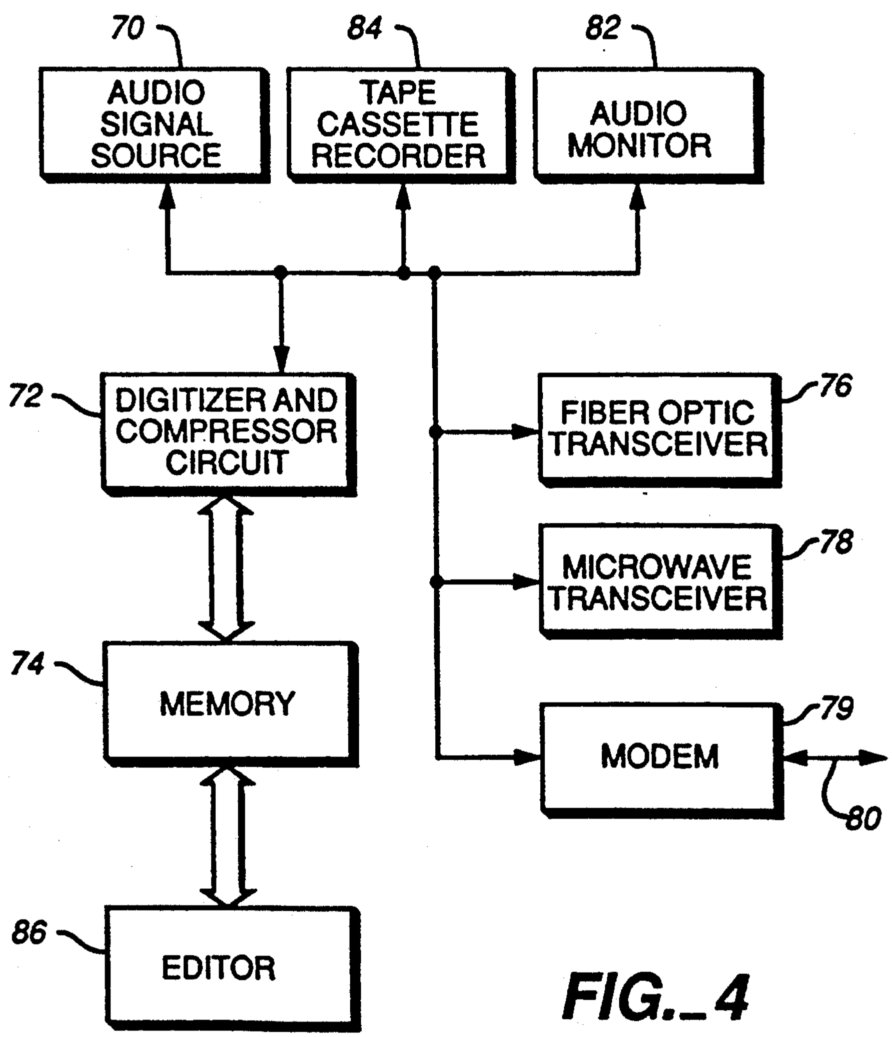

70
84
82
AUDIO SIGNAL SOURCE
TAPE CASSETTE RECORDER
AUDIO MONITOR
72
DIGITIZER AND COMPRESSOR CIRCUIT
FIBER OPTIC TRANSCEIVER
76
MICROWAVE TRANSCEIVER
78
74
MEMORY
MODEM
79
80
86
EDITOR

FIG._4

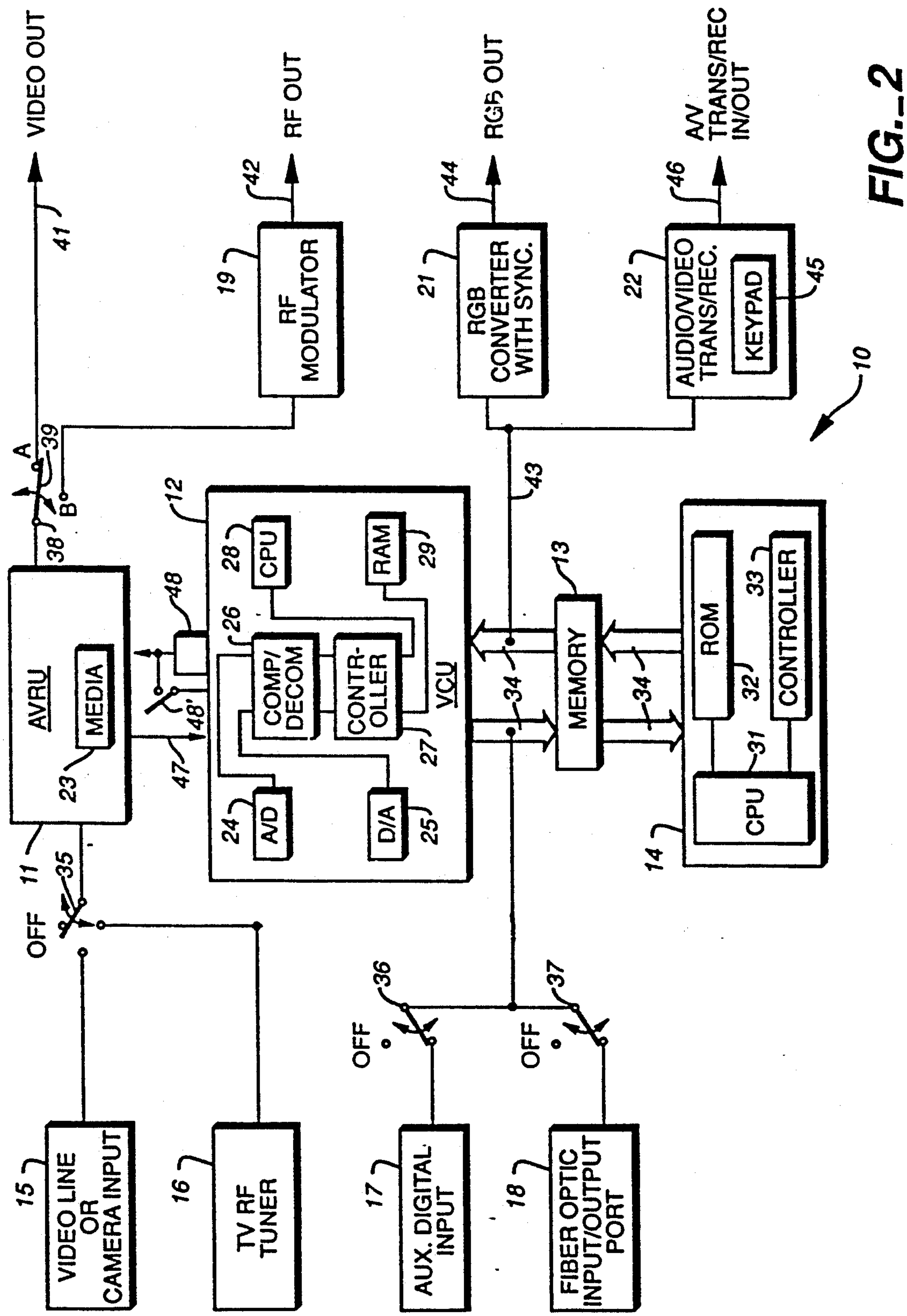
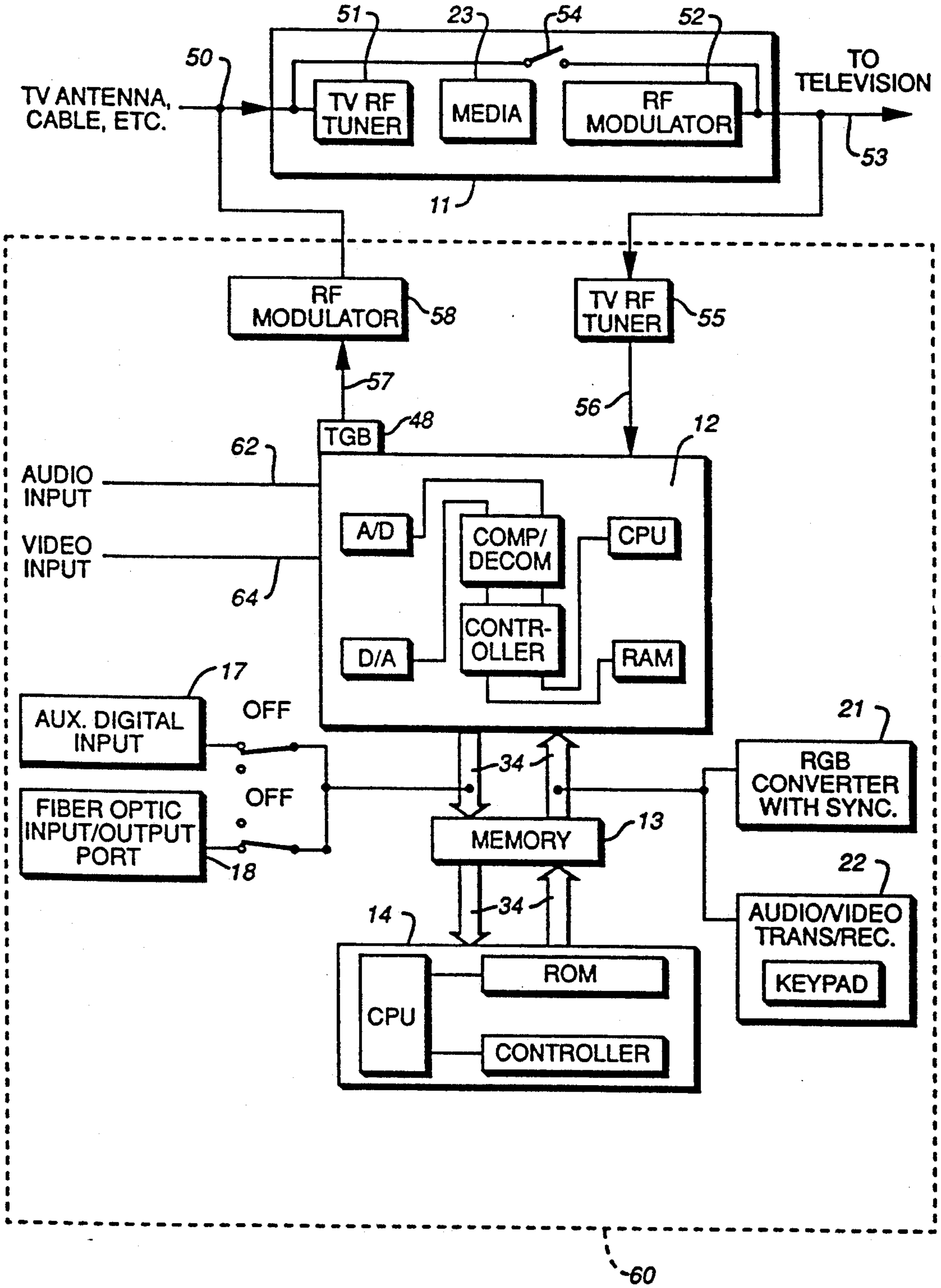

VIDEO LINE OR CAMERA INPUT
TV RF TUNER
AUX. DIGITAL INPUT
FIBER OPTIC INPUT/OUTPUT PORT
OFF
AVRU
MEDIA
VCU
A/D
D/A
COMP/ DECOM
CONTR- OLLER
CPU
RAM
MEMORY
ROM
CONTROLLER
CPU
RF MODULATOR
RGB CONVERTER WITH SYNC.
AUDIO/VIDEO TRANS/REC.
KEYPAD
VIDEO OUT
RF OUT
RGB OUT
A/V TRANS/REC IN/OUT
A
B

FIG._2



TV ANTENNA, CABLE, ETC.
TV RF TUNER
MEDIA
RF MODULATOR
TO TELEVISION
50
51
23
54
52
53
11
RF MODULATOR
58
TV RF TUNER
55
57
56
TGB
48
12
62
AUDIO INPUT
VIDEO INPUT
64
A/D
COMP/ DECOM
CPU
CONTR- OLLER
D/A
RAM
AUX. DIGITAL INPUT
17
OFF
OFF
FIBER OPTIC INPUT/OUTPUT PORT
18
34
MEMORY
13
21
RGB CONVERTER WITH SYNC.
22
AUDIO/VIDEO TRANS/REC.
KEYPAD
14
CPU
ROM
CONTROLLER
60

FIG._3

METHOD FOR HANDLING AUDIO/VIDEO SOURCE INFORMATION

This application is a division of application Ser. No. 07/374,629 filed May 5, 1989 now U.S. Pat. No. 5057932, which is, in turn, a continuation-in-part of application Ser. No. 07/289,776 filed Dec. 27, 1988 now U.S. Pat. No. 4963995.

BACKGROUND OF THE INVENTION

The video cassette recorder (VCR) has added significantly to the usefulness of the home television set. Important or exceptionally good programs may be recorded to be viewed again. Programs appearing at times that are inconvenient for viewing may be recorded for playback at a later time. Recorded movies or other materials, educational or entertaining, may be rented or borrowed for viewing at home. (As used in the remainder of this specification, the term "program" encompasses movies and other types of video and/or audio materials, whether broadcast from a TV station or another source.)

The typical VCR has its own tuner-receiver and a video-recorder. It can receive and record a program from one channel while the television set is being employed to view a program on another channel. Programs are recorded on magnetic tape. The tape is then played back and viewed on the television set. Features commonly included in the VCR are capabilities for advancing the tape forward or backward at a high speed, stopping motion at any frame to hold the image, or simply playing back the recording at normal speed.

Desirable features that are not normally available in a VCR are capabilities for copying recorded programs from one tape or alternative storage medium to a similar or dissimilar storage medium, editing recorded programs and high speed recording. Another desirable, but currently unavailable, feature is the capability for high speed, high quality transmission and reception by optical fiber using

DESCRIPTION OF THE PRIOR ART

U.S. Pat. No. 4,768,110, incorporated herein by reference, describes a VCR having two tape decks included therein. The purpose for the inclusion of two decks rather than the usual single tape deck is to permit the simultaneous viewing of a live RF-modulated TV signal or prerecorded material while recording another live RF-modulated TV signal and to also allow the copying of material from a first magnetic cassette tape onto a second magnetic cassette tape without the use of a second VCR. Viewing of the recorded material during the copying process is also possible in this arrangement. A major disadvantage is that the incorporation of the second tape deck is expensive and limited to magnetic tape, and furthermore, this prior art does not allow for the transmission or reception of recorded material over optical fibers or the high speed reception or transmission of audio/video material in a digital format. An additional disadvantage is the inability for random access editing of the audio/video signal. Furthermore, the additional mechanical structure adds significantly to the overall dimension of the equipment and increases the prospects of mechanical failures.

SUMMARY OF THE INVENTION

In accordance with the invention, an improved audio/video recorder is provided with added features and functions which significantly enhance its usefulness and functionality.

It is, therefore, an object of the present invention to provide an improved audio/video recorder for use in conjunction with an ordinary home television set.

Another object of the invention is to provide in such an improved audio/video recorder a capability for transferring a previously recorded program from one magnetic tape or other storage medium to another.

A further object of the invention is to provide such a capability for transferring a recorded audio/video program without resort to the use of two magnetic tape decks, this being a cumbersome, limited, and expensive approach already proposed in the prior art.

A still further object of the invention is to provide an effective and efficient means for intermediate storage of the audio/video program in digital memory as a means for achieving the transfer of the audio/video program from one tape or storage medium to another.

A still further object of the invention is to provide in such an improved audio/video recorder a capability for accepting various forms of analog or digital audio and video input signals and for converting the analog input signals to digital form when appropriate.

A still further object of the invention is to provide in such an improved audio/video recorder a capability for editing the video input signals without the necessity of using multiple cassettes or recording media.

A still further object of the invention is to provide an improved audio/video recorder for connection to various signal sources including a TV RF tuner, video camera, video and audio line input, and direct audio/video digital input from sources as diverse as a fiber optic input line, a microwave transceiver or a computer.

A still further object of the invention is to provide an improved audio/video recorder having a capability for mixing live audio/video programs with either analog or digital audio/video input signals from another source

A still further object of the invention is to provide an improved audio/video recorder for simultaneously playing, viewing, recording and/or mixing digital and analog audio/video programs from different digital and analog audio/video sources or storage media.

A still further object of the invention is to provide an improved audio/video recorder which maximizes a given storage capacity, through the use of a data compression technique.

A still further object of the invention is to provide an audio/video recorder/transceiver utilizing a data compression technique for efficient storage of data, and efficient transmission and reception of a digitized audio/video program over a telephone line, a fiber optic cable, a microwave transceiver or other data transmission means.

A still further object of the invention is to provide in such an improved audio/video recorder a capability for delivering output signals in different forms or formats including a standard RF modulated output signal for viewing on a television set, a digital output signal for viewing on a high-resolution monitor, and audio output signals for a speaker system.

A still further object of this invention is to provide an improved audio/video recorder which provides for random access to any given segment of a self-stored

audio/video program so that the desired segment may be accessed and viewed without the time-consuming delays normally involved in fast-forward or fast-reverse searching procedures employed in present state-of-the-art VCR's.

A still further object of the invention is to provide an improved audio/video recorder which provides convenience in the editing of stored data by virtue of its random access memory capability.

A still further object of the invention is to provide an improved audio-video recorder which has the potential for enhanced audio and video quality by virtue of its capability for digital audio/video output and digital filtering techniques, and image or audio processing.

Further objects and advantages of the invention will become apparent as the following description proceeds, and the features of novelty which characterize the invention will be pointed out with particularity in the claims annexed to and forming a part of this specification.

BRIEF DESCRIPTION OF THE DRAWING

The present invention may be more readily described with reference to the accompanying drawing, in which:

FIG. **1** is a perspective view of the housing of the audio/video recorder editor/transceiver ("VCR-ET") disclosed and embodying the invention;

FIG. **1A** is an enlarged view of the circled area of FIG. **1**;

FIG. **2** is a functional block diagram of the VCR-ET of FIG. **1**;

FIG. **3** is a functional block diagram of a VCR-ET in accordance with another embodiment of the invention; and

FIG. **4** is a functional block diagram of an audio recorder/transceiver constructed in accordance with the invention.

DESCRIPTION OF THE PREFERRED EMBODIMENT

Referring to the drawing by reference characters, FIGS. **1** and **2** illustrate an improved audio/video recorder editor/transceiver **10** (VCR-ET) comprising an audio/video recording unit (AVRU) **11**, a video control unit (VCU) **12**, memory **13**, digital control unit (DCU) **14**, video line or camera input line **15**, TV RF tuner **16**, auxiliary digital input port **17**, fiber optic input/output port **18**, RF modulator **19**, RGB converter with synchronizer **21**, and an audio/video transmitter/receiver **22** with keypad **45**, all in a common housing.

The audio/video recording unit AVRU **11** may be a video cassette recorder similar to a conventional VCR in which the storage media **23** is a magnetic tape. Alternatively AVRU **11** may operate with other types of storage media including, but not limited to, other magnetic tape formats. AVRU **11** has all the functions of the typical VCR including record, play, rewind, slow motion, fast-forward and single frame hold.

An alternate form of storage media for use in AVRU **11** is the CD-ROM, which is a disk using a derivative of glass or plastic in conjunction with an aluminum or other metallic coating Audio and video signals are stored in the form of irregularities in the aluminum coated surface and are read using a low power laser. In this case, the user would not be able to store or write on the CD-ROM, but would be able to play discs that have been recorded and distributed commercially. The storage of video and audio signals on the CD-ROM is in digital form which is readily accommodated by the video recorder of this invention.

Instead of using a CD-ROM, VCR-ET **10** can use optical discs as media **23**. Such optical discs are similar to a CD-ROM and use a variable power laser to read from or write on the disc.

A first type of optical disc may comprise a WORM (Write Once Read Many) optical disc. This device has the unique capability of writing on the disc permanently. A laser is used to change the magnetic or optical properties of the media. A lower-powered laser is then used to read the data from the disc. Data, in this case, is permanently recorded; it may neither be erased nor written over. A further description of this technology can be found in the November 1988 issue of *The Electronic System Design* magazine (ESD) pages 55–56, incorporated herein by reference.

A second and preferred type of optical disc to be used in AVRU **11** is an erasable optical disc. This disc has full read/write/erase capabilities. With this disc, AVRU **11** has the same record/playback capabilities as a conventional VCR. As an example, erasable optical discs are used in Steven Jobs' "Next" machine as described in *Infoworld*, Volume 10, issue 42, pages 51 and 93, Oct. 17, 1988, incorporated herein by reference. In addition, the random access capabilities of the erasable disc (and of the CD-ROM and WORM) provide additional benefits as will be discussed in a later part of this specification.

A key element of VCR-ET **10**, which is responsible for its improved functionality, is the video control unit or VCU **12** The VCU comprises an analog to digital converter (ADC) **24**, a digital to analog converter (DAC) **25**, a compressor/decompressor **26**, a controller **27**, a central processing unit (CPU) **28** and a random access memory (RAM) **29** VCU **12**, using these elements, accomplishes the digitization and compression of analog signals as well as the reverse process in which the compressed digital signals are decompressed and converted back to analog signals.

As a first step in the processing of the composite video signals within VCU **12**, the sync signals are decoded to isolate signals for each picture frame for processing.

The video signals defining each frame may then be converted to a red analog signal, a green analog signal, and a blue analog signal in a conventional manner. The red, green and blue analog signals are then converted to digital form by the analog to digital converter (ADC) **24**. The frame is divided into a set of closely positioned rows and columns of picture elements or "pixels." Each pixel has a color defined by a set of three digital values defining strength of the primary color components, red, green and blue (RGB) respectively. In one embodiment, each frame is divided into an array of 300 by 300 pixels, with the color and luminance of each pixel being defined by a seven bit word for the red component, a seven bit word for the blue component, and a seven bit word for the green component. These words are generated by ADC **24**. The RGB video signal may also be processed by means of hue-saturation-intensity (HSI) color processing, where appropriate, as described in "Chips for Real-Time Comparisons," *Electronic Engineering Times*, issue 525, Feb. 13, 1989, page 122.

If each frame includes 90,000 pixels (300×300), and each pixel is defined by 21 bits (7 bits per primary color), the digital representation of a single video frame utilizes a sizable block of digital information (i.e., 1.89 megabits/frame) which must be processed very rapidly.

(Approximately 30 frames/second are received from AVRU **11**.) Fortunately the analog to digital conversion of these signals may be accomplished at the desired speed using commercially available analog to digital converter integrated circuits. The analog to digital converter **24** (ADC) is a high-speed, high-accuracy, A to D "flash" converter available as a single IC (integrated circuit). Several different types of such A/D converters are available from Burr-Brown, one of which is the ADC **600**. Part number TIC024, manufactured by Tektronix, Inc. is also appropriate. Other types of devices appropriate for this function are described in an article by K. Rogers entitled "8-bit A/D Flash Hits 500 Msamples", Electronic Engineering Times, Dec. 12, 1988, page 90, incorporated herein by reference.

Compression of the digital data defining a video frame and the reverse process (decompression) are accomplished by compressor/decompressor **26**. Various algorithms may be employed in the compression process which enable the representation of a series of numbers by a reduced number of digits. As an example, compression algorithms like CCITT Group IV may be used.

In one optional embodiment, to further reduce the amount of memory required to store a program, the compression algorithm can simply record data corresponding to only those pixels which change color from one frame to the next This results in considerable memory space savings, since not all pixels change color each frame. Basing calculation upon 10% of the pixels changing from one frame to the next, it is estimated that memory requirements using this technique are cut by about 90%. It is also estimated that on the average, the CCITT Group IV algorithm can cut memory requirements by another 95%. Thus, if no data compression technique is used, it would take approximately 51.03 gigabytes to store a 2 hour video program, but by using the above compression techniques, it is estimated that memory **13** will require only 250 megabytes.

Controller **27** handles timing and aids in the communication between the different elements of VCU **12**, and between VCU **12**, AVRU **11** and memory **13**.

In one embodiment, the audio portion of the program is periodically sampled and digitized by analog to digital conversion. In one embodiment, this is done at a sample rate of 88,000/second, one byte per sample, to yield compact disc quality sound. The sampling rate could be dropped to reduce memory requirements. Also, the audio data can be compressed with conventional algorithms.

The process of converting either from analog to digital or from digital to analog requires memory for intermediate storage. Random Access Memory (RAM) **29** serves in this capacity For this purpose either a DRAM (Dynamic RAM) or a SRAM (static RAM) may be employed An example of a DRAM is the TI (Texas Instruments) TMX4C1024; an example of a SRAM is the INMOS IMS-1203. RAM **29** should have sufficient capacity to store at least two full uncompressed frames (e.g., about 472 KB).

The CPU (Central Processing Unit) **28** is a microprocessor which controls the digitization process of VCU **12**. CPU **28** works with controller **27** to control and communicate with the other elements of the VCU. There are numerous commercially available microprocessors that are appropriate for this application. The Intel 80286, Intel 80386, Motorola 68020, and Motorola 68030 are examples. A more complete description of the microprocessors can be found in the Oct. 27, 1988 issue of *Electronic Design News* (EDN), pages 231 and 242, incorporated herein by reference, or in the applicable data sheets.

Controller **27**, CPU **28** and RAM **29** serve in the same manner during the reverse processes, i.e., decompression and digital to analog conversion. Decompression is first accomplished in compressor/decompressor **26**. The decompressed digital signal is then converted to an analog signal by digital to analog converter (DAC) **24** (assuming its destination requires an analog form). In the course of converting the decompressed signals from the VCU **12** for use by the AVRU **11** the signals are synchronized by the time base generator (TBG) or corrector **48**. TBG generator **48** inserts synchronization pulses into the signal provided by VCU **12** to identify individual raster scan lines and frames so that the resulting signal can be used by a conventional television set or VCR. TBG **48** can be bypassed by shunt switch **48'** for the purpose of transmitting either compressed or decompressed signals from VCU **12** directly to the AVRU **11** in an uncorrected time based mode.

DAC **25** provides the inverse of the function performed by A/D converter **24**. DAC **25** is a high-speed, high accuracy digital to analog converter. An example of such a converter is the Burr-Brown DAC60 digital to analog converter.

Different types of memory technologies are adaptable for use in memory **13**. As mentioned earlier, DRAM and SRAM semiconductor memories are commonly used for applications of this type and are readily available.

One type of random access memory is CMOS (Complimentary Metal Oxide Semiconductor). The CMOS memory has the advantage of a relatively low power requirement and is readily adaptable for use of battery backup for semipermanent data storage. Other types of memory include the above mentioned optical disc memories, bubble memories and magnetic disks. Another appropriate data storage media may be "Digital Paper" available from ICI Image data of Wilmington, Delaware.

Emerging memory technologies may also prove advantageous with capabilities for mass data storage in even smaller physical dimensions.

Digital Control Unit (DCU) **14** comprises a CPU (Central Processor Unit) **31**, a ROM (Read Only Memory) **32** and a controller **32**. DCU **14** is responsible for all of the digital editing processes. Through the use of DCU **14**, video segments may be edited and rearranged. Thus, one may use DCU **14** to rearrange the scenes in a program, alter the program sound track, etc.

In addition, a program may be edited, one frame at a time, by changing the contrast, brightness, sharpness, colors, etc. (Alteration of the contrast, brightness, sharpness and colors can be automated as well.) In one embodiment, images can be rotated, scaled (i.e., made larger or smaller), etc. In addition, pixel by pixel editing can be accomplished by DCU **14**, e.g., in a manner similar to a PC paint program. Similar editing features can be incorporated for the audio portion of each program. In one embodiment, a display such as a flat panel video display (not shown) is built into the VCR-ET. A user interface control panel of DCU **14** allows a user to select a desired frame number from a menu on the display. The VCR-ET then displays a strip of frames (including several frames before and after the selected frame). The user can delete frames in a strip, select a

point where other frames are to be inserted into the program, or edit different frames (i.e., alter contrast, brightness, sharpness, colors, etc.). In one embodiment, a user input device such as a light pen or mouse can be used to select individual frames in a strip for editing.

Instead of incorporating a flat display into VCR-ET **10**, in another embodiment, a television coupled to output lead **42** of RF modulator **19** can be used during editing.

CPU **31** is a microprocessor of the type described in connection with the CPU **28** of VCU **12**. Controller **33** is an integrated circuit which handles the timing and interfacing between DCU **14** and memory **13**. ROM **32** holds the necessary step-by-step editing programs which are installed at the factory A currently available example of a suitable ROM for this application is the Texas Instruments part TMS47256. CPU **31** and controller **33** together control the editing process as they execute the programs stored in ROM **32**.

The VCU **12**, memory **13** and DCU **14** communicate with each other via a high speed data bus **34**. The high speed data bus is required in order to meet bandwidth requirements. Examples of suitable data bus devices are Motorola's VME bus, Intel's Multibus and the Optobuss (U.S. Pat. No. 4,732,446).

A video line or camera input line **15** is provided to enable VCR-ET **10** to receive an input signal from a source such as a television camera, a conventional VCR, a television tuner, or another VCR, etc. The signals received at input line **15** are typically carried by a coaxial cable and are in the form of a standard television composite signal. As used throughout this specification, the words "standard television composite signal" or its acronym STCS shall be read to include any one of the following: NTSC, PAL, SECAM, HDTV, or any American or European broadcast signal standards. (NTSC, PAL and SECAM are discussed in "Reference Data for Radio Engineers", published by Howard W. Sams & Co. in 1983, incorporated herein by reference.) An NTSC composite signal is defined as the analog signal that carries the chrominance (color), luminance (brightness), synchronization (timing) and audio signals that make up the video signals received and displayed by television and video cassette recorders. These four components are combined into one signal by modulating the components in different ways. (Amplitude modulation and phase modulation are examples.) The standard video line signal is such a composite signal and may be received at input line **15** from one of the above-mentioned sources.

TV RF tuner input port **16** also supplies a composite signal as described in regard to video input line **15**. The difference is that this signal is received from an antenna or cable TV coaxial cable. To receive such a signal, tuner **16** is capable of being set or tuned to receive the desired carrier frequency or television channel.

Selector switch **35** is provided to select either video input line **15** or TV RF tuner **16** as an input signal source to AVRU **11**.

Auxiliary digital input port **17** is employed to receive any acceptable digital signal such as computer-generated video signal or as may be supplied by another VCR-ET. This signal, for example, may be an RGB video signal such as that delivered to computer monitors, or it may be a digitized audio signal (As mentioned above, an RGB signal is a signal which communicates the strength of the red, green and blue color components for the pixels that make up each video frame.) Switch **36** selects whether the digital video/audio input signal is chosen from auxiliary digital input port **17**. Switch **36** supplies the selected signal to high speed data bus **34** which carries the signals in digital form.

Fiber optic port **18** incorporates a fiber optic transceiver Port **18** has a capability for transforming fiber optic (light) signals to electrical signals or for transforming electrical signals to fiber optic signals. Port **18** thus provides a capability for two-way communication between high speed data bus **34** and a fiber optic signal line. The incorporation of fiber optic port **18** in the VCR-ET provides a capability for receiving audio/video signals from or delivering audio/video signals to the fiber optic line such as a fiber optic telephone line. The fiber optic line carries digital signals in the form of light waves over great distances with a high degree of accuracy and reliability and at a high speed (e.g., about 200 megabytes/second). The VCR-ET can receive/transmit a video program at an accelerated rate via fiber optic port **18** from/to a variety of sources. For example a video program may be communicated at an accelerated rate from the first VCR-ET to a second VCR-ET in less time than it would take to view the program. Thus, it is not necessary to access the optical fiber for long periods of time in order to transmit a long video program.

It is also envisioned that in the future, a video library may be established which downloads video programs at an accelerated rate via optical fibers to a subscriber's VCR-ET. After downloading, the program may be viewed, stored in memory, edited and/or a hard copy of the program may be made on magnetic tape, optical disk, etc.

Switch **37** is provided to select connection to the fiber optic input/output port **18**. An OFF or open position is provided. The selected signal is delivered to or supplied from high speed data bus **34**.

Analog output signals from AVRU **11** are delivered to the common terminal **38** of a selector switch **39**. When set to position A, switch **39** delivers the output signal of AVRU **11** directly to a video output line **41** as a standard STCS composite signal; when set to position B switch **39** delivers the output of VRU **11** to the input of RF modulator **19**. Modulator **19** converts the video signal to an RF-modulated composite signal for delivery to such devices as televisions and conventional VCR's These types of devices play back the video program on a particular frequency channel (such as channel 4) on the television. Delivery to the television or VCR is via RF output line **42**.

Digital output signals from VCR-ET **10** may be dispatched from high speed data bus **34** via line **43** to input leads of RGB converter **21** and audio/video transmitter/receiver **22**.

RGB converter **21** converts the STCS signal into an RGB signal as required by computer monitors and similar display devices. The converted signal is received by a display device connected to RGB converter output line **44**.

VCR-ET **10** includes audio/video transmitter/receiver **22** which is typically a built-in modem. Advantageously, the modem may be used to communicate an audio/video program over conventional phone lines in a manner similar to that described above with respect to optical fibers. The term modem is derived directly from its functionality as a modulator-demodulator which allows transfer of the audio/video signal in a digital format over the standard telephone line. Modems are

commonly available for computers and are currently available in the form of a single integrated circuit. As an example, Sierra Semiconductor offers a 2400 baud single chip modem under its part number SC111006. Representative manufacturers of these single modem IC's can be found in the Apr. 14, 1988 issue of Engineering Design News (EDN), pages 124–125. Some of these single IC modems have the added capability of generating the tones for dialing a phone number. The destination phone number may be entered by means of an optional keyboard/keypad **45** incorporated in the video recorder **10** of the invention. Output port **46** of transmitter/receiver **22** connects directly to the telephone line.

Also associated with Modem **22** is an auxilliary keyboard **45'** (FIG. **1A**) of buttons for commanding the modem to perform tasks such as starting a transmission over phone lines (**45**_a_), terminating a transmission (**45**_b_), automatic telephone answering to receive transmissions (**45**_c_), using an optional speaker (not shown) to monitor phone lines (**45**_d_), using an optional microphone (not shown) to speak over the phone lines (**45**_e_) and for controlling the baud rate (**45**_f_).

The application and utilization of the VCR-ET may include a number of forms or operating modes. In its first and simplest operating mode, AVRU **11** may be operated in the manner of a conventional VCR with signals from an antenna being received by tuner **16** and recorded directly on media **23** in analog form. At the same time the received program may be viewed on the television screen with the television connected at video output terminal **42**. An optional signal source for this type of operation is the video line or camera input line **15** selectable by switch **35**.

In a second operating mode a program stored in media **23** of AVRU **11** may be played back and viewed on the connected television set.

When it is desired to copy a program from one recording media to another, the recording media holding the desired program is installed in the AVRU. The recording media is then played back with optional viewing on a connected television set or other TV monitor or listening through speakers (as appropriate). As the recording media is played back, the analog signals from the recording media (video and/or audio) are dispatched to VCU **12** via connection **47**. The analog signals are converted to digital signals by ADC **24**, compressed by compressor/decompressor **26** and the compressed digital signals are stored in memory **13**. The foregoing operations are accomplished under the control of controller **27** and CPU **28**. RAM **29** is used for interim data storage during this process. Once the complete video/audio program has been stored in memory **13**, the recording media from which the stored program has just been read is replaced by blank recording media upon which the stored program is to be copied. CPU **28** in cooperation with controller **27** and RAM **29** then executes the decompression and digital to analog conversion of the program stored in memory **13**, decompression taking place in compressor/decompressor **26**, and digital to analog conversion being accomplished by DAC **25**. The resulting analog program is stored on the blank recording media which constitutes media **23** of AVRU **11**.

In an alternate mode of operation, the decompression circuitry of VCU **12** can be bypassed. Thus, a user has the option of downloading the stored program from memory **13** onto recording media **23** in compressed digital format. The user can then reload the program from media **23** into memory **13** at a future time for viewing, editing or recording back onto recording media **23** in analog form. This capability allows the user to quickly clear memory **13** for other interim uses and also provides the user with a hard copy of the program in digital format. The hard copy in compressed digital format has a number of uses, e.g. it could be archived for later viewing, transmitted by an appropriate independent transmitter, etc.

During the foregoing procedures, DCU **14** may be utilized for editing operations. As the program is being read from the first or original recording media, it is simultaneously viewed on the TV screen, or listened to by means of an audio monitor, converted to digital signals, compressed and stored in memory **13**. Once the digital audio/video program is stored in memory **13**, editing is accomplished by the user through control of DCU **14**, by means of a control panel (not shown) coupled to DCU **14**. If desired, additional audio/video signals may be simultaneously entered into memory **13** and added to those received from VCU **12**. The additional signals may be introduced from auxiliary digital input port **17** or from fiber optic input/output port **18** and may comprise video captions for super imposed position upon the stored video images, or they may be audio commentaries to be added to silent video presentations. In addition, as mentioned above, the order in which various segments appear in the video programs may be altered. Certain undesired segments, such as TV commercials, may be removed. This editing operation is accomplished under the control of DCU **14**.

In still another operating mode, a program stored in media **23** of AVRU **11** or being received by AVRU **11** line **15** (as from a video camera) may be digitized and compressed by VCU **12** and routed via bus **34**, to memory **13**. The data from memory **13** is then routed to line **43**, transmitter/ receiver **22** and to a telephone line. At the other end of the telephone line the signals received are processed by another VCR-ET.

Once received in the second VCR-ET's memory **13**, the digitized program can then either be viewed directly from memory or transferred to storage medium **23**, either in its entirety or in random segments, based on user preference.

In the case of video camera input at input **15** the transmitted signals may comprise a live transmission. Alternatively the transmitted program may be derived from a program stored in media **23** of AVRU **11**. In this case the stored analog program is again decoded, digitized, compressed and transmitted via bus **34** to memory **13**. The data in memory **13** is then communicated via line **43** and transmitter/receiver **22** to telephone lines.

It follows, of course, that digitized video and audio signals from the remote VCR-ET at the other end of the telephone line may be received at line **46**, entered into memory **13** via transmitter/receiver **22**, converted to analog signals by VCU **12**, and recorded on media **23** and then viewed, if desired, on a television set connected at output **42**. A hard copy of the program may also be made for later viewing.

As mentioned earlier, when any of the foregoing operations entail the processing of unmodulated video signals, such signals must first be processed by RF modulator **19** before they can be accepted by devices such as a conventional VCR or television set; when the monitoring means is a computer monitor or a similar display device the signals are processed by RGB converter **21**.

All of the foregoing operations are performed with enhanded quality and efficiency by virtue of the digital, rather than analog, storage and transmission modes and the compressed data storage mechanism, with additional advantages of improved cost and reliability afforded in the case of tape to tape (or other media to media) program transfers by virtue of the requirement for only a single tape deck or other storage device.

FIG. **3** illustrates an alternative embodiment invention in which AVRU **11** is not integral with VCU **12**, memory **13** or editor **14**. In this embodiment, AVRU **11** is a conventional, commercially available VCR which receives a modulated video input signal on an input cable **50**. In this embodiment AVRU **11** includes a RF tuner **51** for demodulating the input signal so it can be stored in media **23**. AVRU **11** also includes a RF modulator **52** for modulating the signal received from media **23** and providing the RF modulated output signal on an output cable **53**, which can be coupled to a television set. (TV RF tuner **51** and RF modulator **52** are provided in typical commercially available VCR's.) A switch **54** is provided to couple input cable **50** to output cable **53** when media **23** is not serving as a video signal source. The VCR-ET of this embodiment includes a TV RF tuner **55** which receives and demodulates the signal on cable **53**, and provides the resultant analog audio/video signal on a lead **56**, which is digitized and compressed as described above. In this alternative embodiment, the digitized compressed signal may be processed as described above, e.g. stored in memory **13** (via high speed bus **34**), edited, transmitted by the fiber optic port **18** to another VCR-ET, etc. When it is desired to view a program stored in memory **13**, data from memory **13** is decompressed and converted to an analog signal by VCU **12**, and the resulting signal is provided on an output lead **57** to a RF modulator **58**, which modulates the video signal so that it can be received and stored by AVRU **11** or viewed on a television coupled to cable **53**. (As mentioned above, in the FIG. **3** embodiment, AVRU **11** is a conventional VCR.)

One advantage of the embodiment of FIG. **3** is that many people already own VCR's. Rather than buying apparatus which duplicates much of the hardware already present in their VCR, the embodiment of FIG. **3** would provide to owners of conventional VCR's capabilities which are otherwise currently unavailable in an economical manner.

In one embodiment, analog auxillary audio and video input terminals **62**, **64** are provided so that analog signals may be provided by alternate sources to VCU **12**.

The embodiments described above include means for transmitting/receiving video programs over fiber optic cables However, in an alternative embodiment, either in place of fiber optic port **18** or in addition to fiber optic port **18**, means are provided for transmitting and/or receiving a video program via microwave. In conventional microwave technology, satellite systems and microwave transmitters transmit data using a low power/-high frequency signal. In an embodiment of the invention designed to receive microwaves, the VCR-ET includes an amplifier for amplifying the microwave signal and a demodulator for obtaining the video program signal from the microwave signal. Receiving, amplifying and demodulating the microwave signal can be accomplished with conventional microwave transceiver equipment. The video program signal is typically in digital form, and may be stored, viewed or edited as in the above-described embodiments. Program data from memory **13** can also be transmitted by the microwave transceiver, thereby providing the capability for microwave transmission of stored video programs in compressed digital format. Thus, the invention can be used to receive and transmit programs via microwaves at an accelerated rate similar to and at least as fast as, the transmission and reception of programs over optical fibers. This feature allows transmission and reception of programs in a few minutes or seconds using currently available technology. Both point-to-point microwave transceivers and satellite transceivers may be used.

The embodiments described include means for receiving, storing and transmitting both audio and video signals. However, the invention encompasses apparatus which can store and transmit video signals only and apparatus which can store and transmit audio signals only. An embodiment designed to store and compress audio signals is illustrated in FIG. **4**. Referring to FIG. **4**, an audio signal source **70** (a tape recorder, microphone, record player, etc.) is coupled to a digitizer and compressor circuit **72**, which converts the analog signal to a digital signal and compresses the digital signal in a manner similar to VCU **12** described above. The digital compressed signal can then be stored in a memory **74**. Of importance, data from memory **74** can be transmitted by a fiber optic transceiver **76**, or by a microwave transceiver **78** at an accelerated rate. This is important not only in a home entertainment application, but in other applications as well. For example, a user can dictate an audio presentation and send it to a remote location (e.g. an office) at an accelerated rate without having to monopolize the transmission medium (e.g. the fiber optic cable) for an extended length of time.

The business uses of the embodiment illustrated in FIG. **4** makes home offices feasible for many workers now confined to more traditional offices and also opens new possibilities to business people who are traveling.

In the embodiment of FIG. **4**, data can also be loaded from memory **74**, via a modem **79** over a conventional phone line **80**. Data can also be received from phone line **80**, fiber optic transceiver **76** or microwave transceiver **78**, loaded into memory **74**, and converted to an analog signal by circuit **72**, to be listened to via an audio monitor **82**, or to be stored on an audio tape cassette **84** or other storage media.

An editor **86** is optionally provided so that the data in memory **74** may be edited, e.g., by rearranging the order of portions of the audio program, increasing or decreasing the volume of portions (or different frequency components) of the audio program, or enhancing the audio program through filtering techniques (e.g. to remove static and noise).

An improved audio/video recorder with significantly expanded functional capabilities is thus provided in accordance with the stated objects of the invention and although but a single embodiment of the invention has been illustrated and described, it will be apparent to those skilled in the art that various changes and modifications may be made therein without departing from the spirit of the invention or from the scope of the appended claim. For example, the VCR-ET can be constructed so as to be portable Thus, it could be carried to a location where it is desired to record a program, and used to edit the program after it is recorded with a video camera. Other modifications will be apparent to those skilled in the art in light of the present specification.

What is claimed is:

1. A method for handling audio/video source information, the method comprising:

receiving audio/video source information;

compressing the received audio/video source information into a time compressed representation thereof having an associated burst time period that is shorter than a time period associated with a real time representation of the received audio/video source information;

storing said time compressed representation of the received audio/video source information; and

transmitting, in said burst time period, the stored time compressed representation of the received audio/video source information to a selected destination.

2. A method as in claim 1 further comprising the steps of:

editing the stored time compressed representation of said audio/video source information; and

storing the edited time compressed representation of said audio/video source information.

3. A method as in claim 2 further comprising the step of monitoring the stored, time compressed representation of said audio/video source information during editing.

4. A method as in claim 1 wherein the step of transmitting comprises transmitting said time compressed representation of said audio-video source information over an optical channel.

5. A method as in claim 1 wherein the step of transmitting comprises transmitting said time compressed representation of said audio-video source information over a telephone transmission channel.

6. A method as in claim 1 wherein the step of storing comprises storing the time compressed representation of said audio/video source information on an optical disk.

7. A method as in claim 1 wherein the step of storing comprises storing the time compressed representation of said audio/video source information in a semiconductor memory.

8. A method as in claim 1 wherein:

said audio/video source information comprises analog audio/video source information;

said method further comprises the step of converting said analog audio/video source information to corresponding digital audio/video source information;

said step of compressing comprises compressing said corresponding digital audio/video source information into a digital time compressed representation thereof having an associated burst time period that is shorter than a time period associated with a real time representation of said digital audio/video source information; and

said step of storing comprises storing said digital time compressed representation of said corresponding digital audio/video source information.

9. A method as in claim 1 wherein:

said audio/video source information comprises digital audio/video source information;

said step of compressing comprises compressing said digital audio/video source information into a digital time compressed representation thereof having an associated burst time period that is shorter than a time period associated with a real time representation of said digital audio/video source information; and

said step of storing comprises storing said digital time compressed representation of said digital audio/video source information.

10. A method as in claim 8 wherein said audio/video source information comprises information received from a television camera.

11. A method as in claim 8 wherein said audio/video source information comprises information received from an analog video tape recorder.

12. A method as in cl aim 8 wherein said audio/video source information comprises information received from a television RF tuner.

13. A method as in claim 8 wherein said audio/video source information comprises information transmitted by a remotely located television transmitter.

14. A method as in claim 8 wherein said audio/video source information comprises information received from a cable television system.

15. A method as in claim 9 wherein said audio/video source information comprises information received from a computer.

16. A method as in claim 9 wherein said audio/video source information comprises information received over a fiber optic transmission line.

17. A method for handling audio/video source information, the method comprising:

receiving audio/video source information as a time compressed representation thereof, said time compressed representation of said audio/video source information being received over an associated burst time period that is shorter than a real time period associated with real time playback of said audio/video source information;

storing the time compressed representation of said received audio/video source information; and

transmitting, in said burst time period, the stored time compressed representation of said received audio/video source information to a selected destination.

18. A method as in claim 17 wherein said audio/video source information comprises information received over an optical channel from a video library storing a multiplicity of programs of audio/video source information as time compressed representations thereof for selective retrieval by a user in an associated burst time period.

19. A method as in claim 17 wherein said audio/video source information comprises information received over a communications link from a video library storing a multiplicity of programs of audio/video source information as time compressed representations thereof for selective retrieval by a user in an associated burst time period.

20. A method as in claim 1 further comprising the steps of:

selectively decompressing the stored time compressed representation of said audio/video source information;

editing the selectively decompressed time compressed representation of said audio/video source information; and

storing the edited selectively decompressed time compressed representation of said audio/video source information.

21. A method as in claim 1 further comprising the steps of:

selectively decompressing the stored time compressed representation of said audio/video source information;

editing the selectively decompressed time compressed representation of said audio/video source information;

recompressing the edited selectively decompressed time compressed representation of said audio/video source information; and

storing the recompressed edited selectively decompressed time compressed representation of said audio/video source information.

22. A method as in claim 1 further comprising the steps of:

selectively decompressing the stored time compressed representation of said audio/video source information; and

visually displaying the selectively decompressed time compressed representation of said audio/video source information for viewing by a user.

23. A method as in claim 8 further comprising the steps of:

selectively decompressing the stored digital time compressed representation of said corresponding digital audio/video source information;

editing the selectively decompressed digital time compressed representation of said corresponding digital audio/video source information; and

storing the edited selectively decompressed digital time compressed representation of said corresponding digital audio/video source information.

24. A method as in claim 23 further comprising the step of visually displaying the selectively decompressed digital time compressed representation of said corresponding digital audio/video source information for selective viewing by a user during editing.

25. A method as in claim 8 further comprising the steps of:

selectively decompressing the stored digital time compressed representation of said corresponding digital audio/video source information; and

visually displaying the selectively decompressed digital time compressed representation of said corresponding digital audio/video source information for selective viewing by a user.

26. A method as in claim 9 further comprising the steps of:

selectively decompressing the stored digital time compressed representation of said digital audio/video source information;

editing the selectively decompressed digital time compressed representation of said digital audio/video source information; and

storing the edited selectively decompressed digital time compressed representation of said digital audio/video source information.

27. A method as in claim 26 further comprising the step of visually displaying the selectively decompressed digital time compressed representation of said digital audio/video source information for selective viewing by a user during editing.

28. A method as in claim 9 further comprising the steps of:

selectively decompressing the stored digital time compressed representation of said digital audio/video source information; and

visually displaying the selectively decompressed digital time compressed representation of said digital audio/video source information for selective viewing by a user.

29. A method as in claim 8 wherein said analog audio/video source information is received from a video tape recorder.

30. A method for handling audio/video source information, the method comprising:

providing a network that includes a plurality of audio/video transceivers, coupled via one or more communication links;

receiving audio/video source information at one or more of said plurality of audio/video transceivers;

compressing said received audio/video source information into a time compressed representation thereof having an associated burst time period that is shorter than a time period associated with a real time representation of said received audio/video source information;

storing the time compressed representation of the received audio/video source information; and

transmitting, in said burst time period, the stored time compressed representation of the received audio/video source information to one or more of said plurality of audio/video transceivers.

31. A method as in claim 30 wherein said audio/video source information is received over one or more optical transmission channels and the stored time compressed representation of the received audio/video source information is transmitted over one or more optical transmission channels.

32. A method as in claim 30 wherein the stored time compressed representation of the received audio/video source information is transmitted over one or more telephone transmission channels.

33. A method as in claim 30 wherein the time compressed representation of the received audio/video source information is stored in an optical disk memory.

34. A method as in claim 30 wherein the time compressed representation of the received audio/video source information is stored in a semiconductor memory.

35. A method as in claim 30 wherein one of said plurality of audio/video transceivers stores a library containing a multiplicity of programs of audio/video source information as time compressed representations thereof for selective transmission, in an associated burst time period, to one or more of the remaining plurality of audio/video transceivers.

36. A method as in claim 30 further comprising the step of recording the stored time compressed representation of said audio/video source information onto a removable recording medium.

37. A method as in claim 30 further comprising the steps of:

decompressing the stored time compressed representation of said audio/video source information; and

recording the decompressed time compressed representation of said audio/video source information onto a removable storage medium.

38. A method as in claim 36 wherein the stored time compressed representation of said audio/video source information is recorded onto a magnetic tape with a video tape recorder.

39. A method as in claim 37 wherein the stored time compressed representation of said audio/video source information is recorded onto a magnetic tape within a video tape recorder.

40. A method as in claim 36 wherein the stored time compressed representation of said audio/video source information is recorded onto one or more write-once read-many (WORM) optical disks within an optical disk drive.

41. A method as in claim 37 wherein the stored time compressed representation of said audio/video source information is recorded onto one or more write-once read-many (WORM) optical disks within an optical disk drive.

42. A method as in claim 36 wherein the stored time compressed representation of said audio/video source information is recorded onto one or more erasable optical disks within an optical disk drive.

43. A method as in claim 37 wherein the stored time compressed representation of said audio/video source information is recorded onto one or more erasable optical disks within an optical disk drive.

44. A method as in claim 1 further comprising the step of recording the stored time compressed representation of said audio/video source information onto a removable recording medium.

45. A method as in claim 2 further comprising the step of recording the edited time compressed representation of said audio/video source information onto a removable recording medium.

46. A method as in claim 45 further comprising the step of visually displaying the time compressed representation of said audio/video source information stored on said removable recording medium for selective viewing by a user.

47. A method as in claim 17 further comprising the step of recording the time compressed representation of said audio/video source information onto a removable recording medium.

48. A method as in claim 20 further comprising the step of recording the edited decompressed time compressed representation of said audio/video source information onto a removable recording medium.

49. A method as in claim 1 further comprising the steps of:

selectively decompressing the stored time compressed representation of said audio/video source information; and

recording the selectively decompressed time compressed representation of said audio/video source information onto a removable recording medium.

50. A method as in claim 22 further comprising the steps of:

recording the selectively decompressed time compressed representation of said audio/video source information onto a removable recording medium; and

visually displaying the selectively decompressed time compressed representation of said audio/video source information for viewing by a user.

51. A method as in claim 9 wherein said digital audio/video source information is received from a CD-ROM.

52. A method as in claim 9 wherein said digital audio/video source information is received from an erasable optical disk.

53. A method as in claim 17 wherein said audio/video source information comprises information received from a television RF tuner.

54. A method as in claim 1 further comprising the step of recording the stored time compressed representation of said audio/video source information onto a magnetic recording medium.

55. A method as in claim 2 further comprising the step of recording the stored edited time compressed representation of said audio/video source information onto a magnetic recording medium.

56. A method as in claim 17 further comprising the step of recording the stored time compressed representation of said audio/video source information onto a magnetic recording medium.

57. A method as in claim 20 further comprising the step of recording the edited decompressed time compressed representation of said audio/video source information onto a magnetic recording medium.

58. A method as in claim 1 further comprising the steps of:

selectively decompressing the stored time compressed representation of said audio/video source information; and

recording the selectively decompressed stored time compressed representation of said audio/video source information onto a magnetic storage medium.

59. A method as in claim 22 further comprising the step of recording the selectively decompressed time compressed representation of said audio/video source information onto a magnetic recording medium.

60. A method for handling analog and/or digital audio/video source information, the method comprising the steps of:

receiving analog and/or digital audio/video source information;

converting received analog audio/video source information to corresponding digital audio/video source information;

converting received digital audio/video source information to corresponding analog audio/video source information;

compressing said received digital or converted corresponding digital audio/video source information into a time compressed representation thereof having an associated burst time period that is shorter than a time period associated with a real time representation of said received digital or converted corresponding digital audio/video source information;

storing said time compressed representation;

decompressing said time compressed representation into a real time representation of said received digital or converted corresponding digital audio/video source information;

storing said real time representation; and

transmitting said time compressed representation to a selected destination.

61. A method as in claim 60 further comprising the step of supplying timing information for association with said time compressed representation.

62. A method as in claim 60 further comprising the step of recording said received analog or corresponding analog audio/video source information onto a recording medium.

63. A method as in claim 60 further comprising the step of recording said received digital or corresponding digital audio/video source information onto a recording medium.

64. A method as in claim 62 wherein said received analog or corresponding analog audio/video source information is recorded onto a magnetic tape recording medium.

65. A method as in claim 63 wherein said received digital or corresponding digital audio/video source information is recorded onto a magnetic tape recording medium.

66. A method as in claim 63 wherein said received digital or corresponding digital audio/video source information is recorded onto a CD-ROM.

67. A method as in claim 63 wherein said received digital or corresponding digital audio/video source information is recorded onto a WORM optical disk.

68. A method as in claim 63 wherein said received digital or corresponding digital audio/video source information is recorded onto an erasable optical disk.

69. A method as in claim 60 wherein said received analog and/or digital audio/video source information is received from an audio/video recording and playback apparatus.

70. A method as in claim 60 wherein said digital audio/video source information is received over a high speed bus.

71. A method as in claim 60 wherein said digital audio/video source information is received over an optical bus.

72. A method as in claim 60 further comprising the step of selectively editing the received analog and/or digital audio/video source information.

73. A method for handling audio/video source information, the method comprising:

receiving audio/video source information comprising a multiplicity of video frames in the form of one or more full motion video programs;

compressing said received audio/video source information into a time compressed representation thereof having an associated burst time period that is shorter than a time period associated with a real time representation of said received audio/video source information;

storing the time compressed representation of said received audio/video source information; and

transmitting, over a microwave channel, in said burst time period, the stored time compressed representation of said received audio/video source information to a selected destination.

74. A method for handling audio/video source information, the method comprising:

receiving audio/video source information comprising a multiplicity of video frames in the form of one or more full motion video programs;

compressing said received audio/video source information into a time compressed representation thereof having an associated burst time period that is shorter than a time period associated with a real time representation of said received audio/video source information;

storing the time compressed representation of said received audio/video source information in a bubble memory; and

transmitting, in said burst time period, the stored time compressed representation of said received audio/video source information to a selected destination.

75. A method for handling audio/video source information, the method comprising:

receiving audio/video source information comprising a multiplicity of video frames in the form of one or more full motion video programs;

compressing said received audio/video source information into a time compressed representation thereof having an associated burst time period that is shorter than a time period associated with a real time representation of said received audio/video source information;

storing the time compressed representation of said received audio/video source information in a digital paper memory; and

transmitting, in said burst time period, the stored time compressed representation of said received audio/video source information to a selected destination.

76. A method for handling audio/video source information, the method comprising:

receiving audio/video source information comprising a multiplicity of video frames in the form of one or more full motion video programs;

compressing said received audio/video source information into a time compressed representation thereof having an associated burst time period that is shorter than a time period associated with a real time representation of said received audio/video source information;

storing the time compressed representation of said received audio/video source information on one or more magnetic disks; and

transmitting, in said burst time period, the stored time compressed representation of said received audio/video source information to a selected destination.

77. A method for handling audio/video source information, the method comprising:

receiving audio/video source information as a time compressed digital representation thereof, said audio/video source information comprising a multiplicity of video frames in the form of one or more full motion video programs selected from a video library storing a multiplicity of full motion video programs in a time compressed digital representation thereof for selective retrieval, said time compressed digital representation of the received audio/video source information being received in an associated burst time period that is shorter than a time period associated in an associated burst time period that is shorter than a time period associated with a real time representation of said received audio/video source information;

storing the time compressed digital representation of said received audio/video source information; and

transmitting, in said burst time period, the stored time compressed digital representation of said received audio/video source information to a selected destination.

* * * * *

UNITED STATES PATENT AND TRADEMARK OFFICE

CERTIFICATE OF CORRECTION

PATENT NO. : 5,164,839

DATED : November 17, 1992

INVENTOR(S) : Richard A. Lang

It is certified that error appears in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

Column 1, line 43, after "using" insert "the VCR";

Column 3, line 62, after "coating" insert a period;

Column 4, line 35, after "29" insert a period;

Column 7, line 65, after "signal" insert a period;

Column 8, line 6, after "ceiver" insert a period;

Column 8, line 47, after "VCR's" insert a period;

Column 16, line 63, "with" should be "within";

Column 20, line 52, cancel "in an associated burst time"; and

Column 20, line 53, cancel the entire line

Signed and Sealed this

Sixteenth Day of November, 1993

Attest:

Bruce Lehman

BRUCE LEHMAN

Attesting Officer

Commissioner of Patents and Trademarks

EXHIBIT C

US005995705A

United States Patent [19]

Lang

[11] **Patent Number: 5,995,705**

[45] **Date of Patent: *Nov. 30, 1999**

[54] **BURST TRANSMISSION APPARATUS AND METHOD FOR AUDIO/VIDEO INFORMATION**

[75] Inventor: **Richard A. Lang**, Cave Creek, Ariz.

[73] Assignee: **Instant Video Technologies, Inc.**, San Francisco, Calif.

[*] Notice: This patent is subject to a terminal disclaimer.

[21] Appl. No.: **08/896,727**

[22] Filed: **Jul. 18, 1997**

Related U.S. Application Data

[63] Continuation of application No. 08/624,958, Mar. 28, 1996, abandoned, which is a continuation of application No. 07/976,542, Nov. 16, 1992, abandoned, which is a division of application No. 07/775,182, Oct. 11, 1991, Pat. No. 5,164,839, which is a continuation-in-part of application No. 07/289,776, Dec. 27, 1988, Pat. No. 4,963,995.

[51] **Int. Cl.**[6] **H04N 5/76**

[52] **U.S. Cl.** **386/46**; 386/109

[58] **Field of Search** 386/46, 52, 101, 386/109, 96, 106, 112; 348/384; H04N 5/76

[56] **References Cited**

U.S. PATENT DOCUMENTS

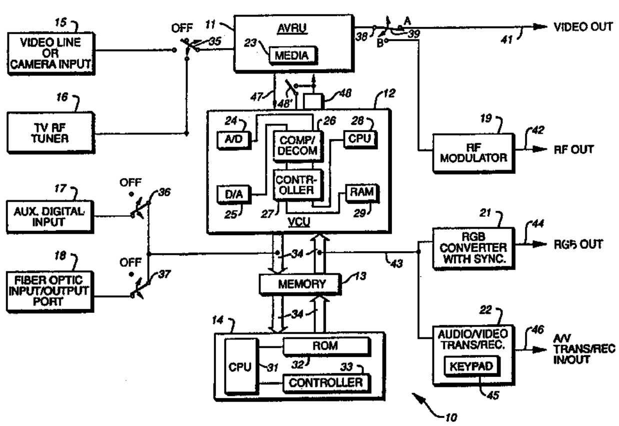
2,987,614	6/1961	Roberts et al.	250/6
4,179,709	12/1979	Workman	353/133
4,300,161	11/1981	Haskell	358/142
4,400,717	8/1983	Southworth et al.	358/13
4,446,490	5/1984	Hoshimi et al.	360/32
4,467,473	8/1984	Arnon et al.	370/109
4,506,387	3/1985	Walter	455/612
4,511,934	4/1985	Ohira et al.	360/55
4,516,156	5/1985	Fabris et al.	358/85
4,521,806	6/1985	Abraham	358/86
4,563,710	1/1986	Baldwin	360/9.1
4,625,080	11/1986	Scott	379/104
4,654,484	3/1987	Reiffel et al.	379/53
4,698,664	10/1987	Nichols et al.	358/10
4,709,418	11/1987	Fox et al.	455/612
4,724,491	2/1988	Lambert	358/310
4,736,239	4/1988	Sprague et al.	358/21 R
4,743,959	5/1988	Frederiksen	358/11
4,750,034	6/1988	Lem	358/84
4,768,110	8/1988	Dunlap et al.	360/33.1
4,774,574	9/1988	Daly et al.	358/133
4,785,349	11/1988	Keith et al.	358/136
4,821,208	4/1989	Ryan et al.	364/518
4,829,372	5/1989	McCalley et al.	348/7
4,851,931	7/1989	Parker et al.	360/15
4,868,653	9/1989	Golin et al.	358/133
4,888,648	12/1989	Takeuchi et al.	358/335
4,891,694	1/1990	Way	348/7
4,897,717	1/1990	Hamilton	358/133
4,918,523	4/1990	Simon et al.	358/133
4,920,432	4/1990	Eggers et al.	348/8
4,941,054	7/1990	Muramoto	358/310
4,943,865	7/1990	Hales et al.	358/335
4,963,995	10/1990	Lang	358/335
4,974,178	11/1990	Izeki et al.	364/523
4,987,552	1/1991	Nakamura	358/335
5,006,936	4/1991	Hooks, Jr.	358/335
5,057,932	10/1991	Lang	358/335
5,068,733	11/1991	Bennett	348/7
5,164,839	11/1992	Lang	358/335
5,220,420	6/1993	Hoarty et al.	358/86

Primary Examiner—Huy Nguyen
Attorney, Agent, or Firm—Carr & Ferrell LLP

[57] **ABSTRACT**

An improved video recorder/transceiver with expanded functionality ("VCR-ET") including a capability for storing video and video programs in digital format, editing such programs, transferring such programs onto a hard copy magnetic media, and transmitting such programs to a remote location using a second VCR-ET. The increased functionality is realized through the use of analog to digital conversion, signal compression and intermediate storage in an integrated circuit, random access memory. The recorder/transmitter has capabilities to transmit and receive program information in either a compressed or decompressed format over fiber optic lines, conventional phone lines or microwaves.

24 Claims, 4 Drawing Sheets



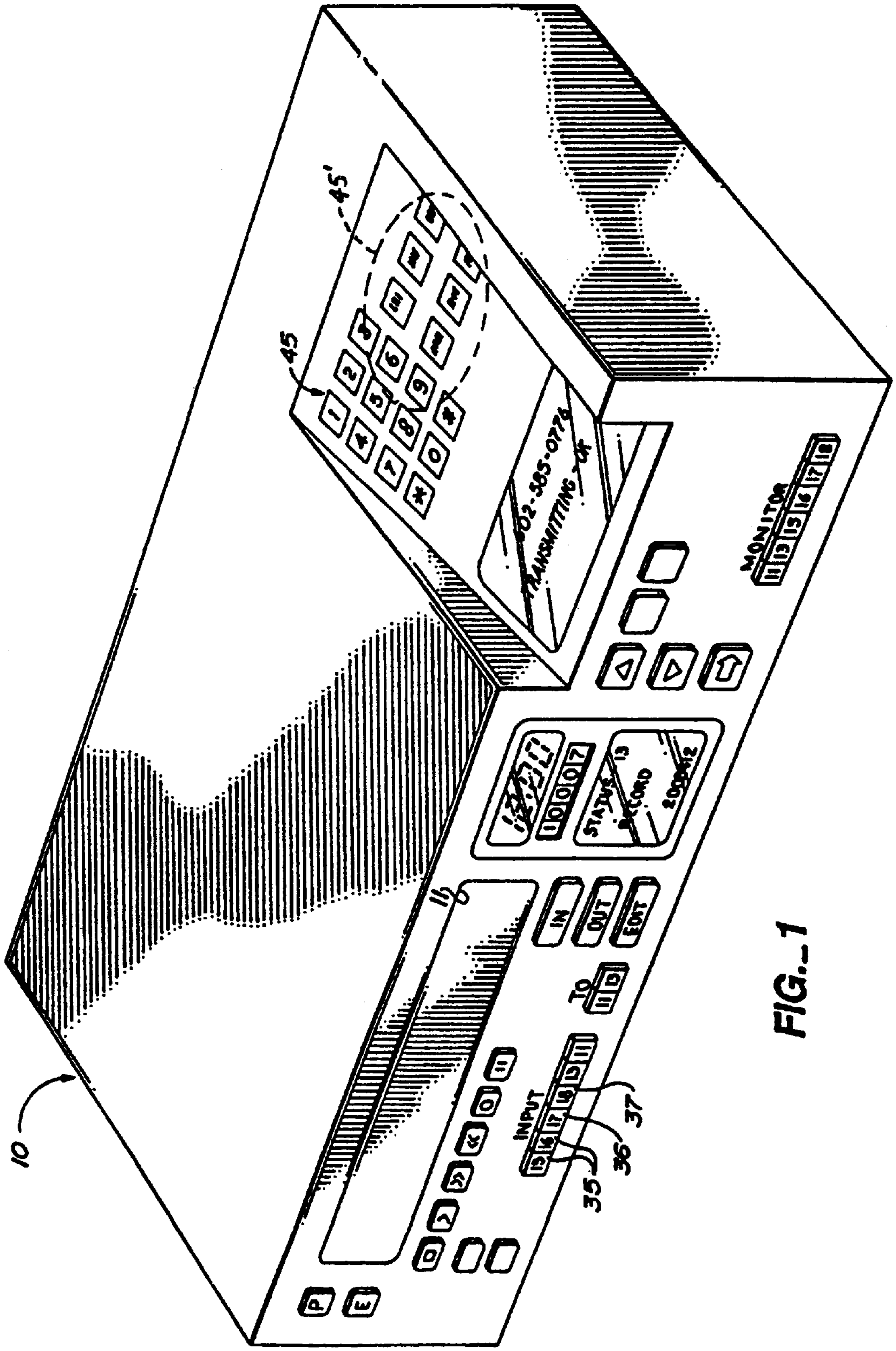

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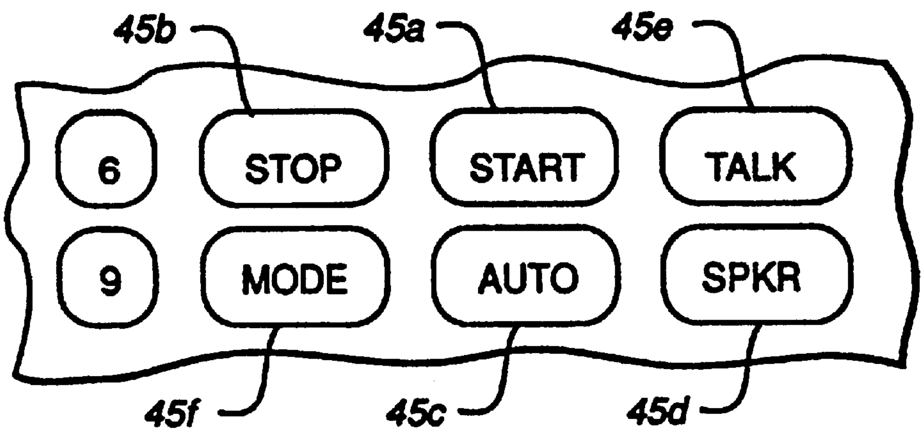
FIG._1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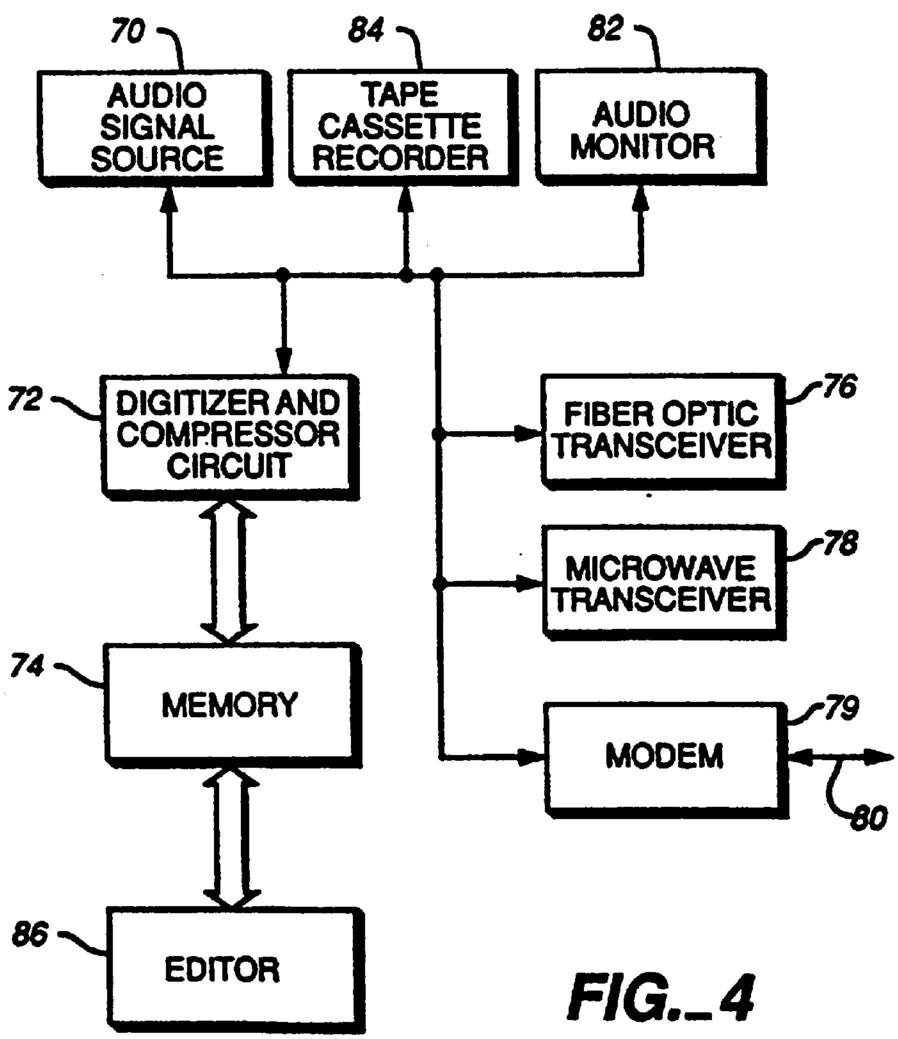
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AUDIO SIGNAL SOURCE
TAPE CASSETTE RECORDER
AUDIO MONITOR
72
DIGITIZER AND COMPRESSOR CIRCUIT
FIBER OPTIC TRANSCEIVER
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MICROWAVE TRANSCEIVER
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74
MEMORY
MODEM
79
80
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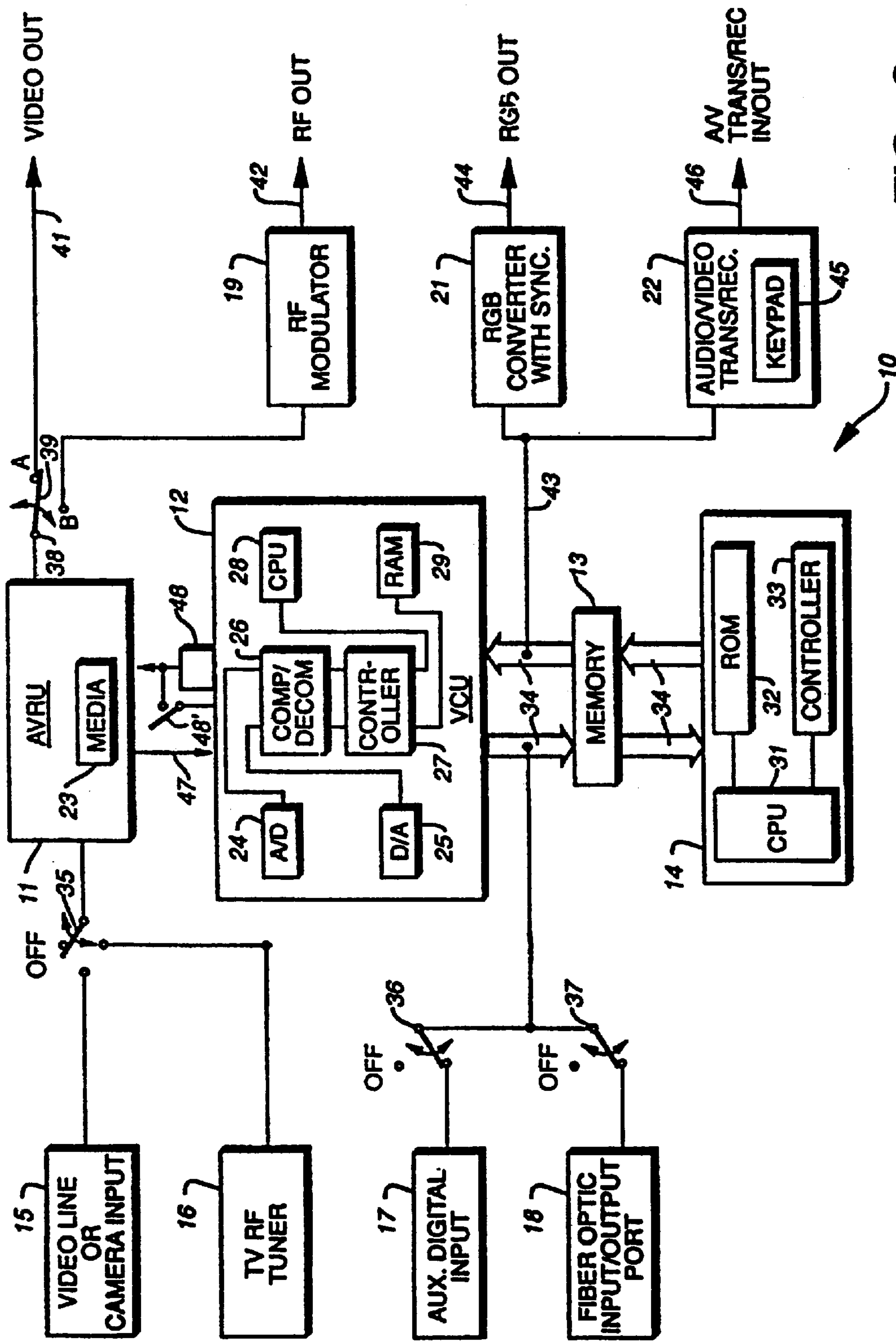
EDITOR

FIG._4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RF OUT
RGB OUT
A/V TRANS/REC IN/OUT
RF MODULATOR
RGB CONVERTER WITH SYNC.
AUDIO/VIDEO TRANS/REC.
KEYPAD
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CPU
RAM
COMP/ DECOM
CONTR- OLLER
VCU
A/D
D/A
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ROM
CONTROLLER
CPU
VIDEO LINE OR CAMERA INPUT
TV RF TUNER
AUX. DIGITAL INPUT
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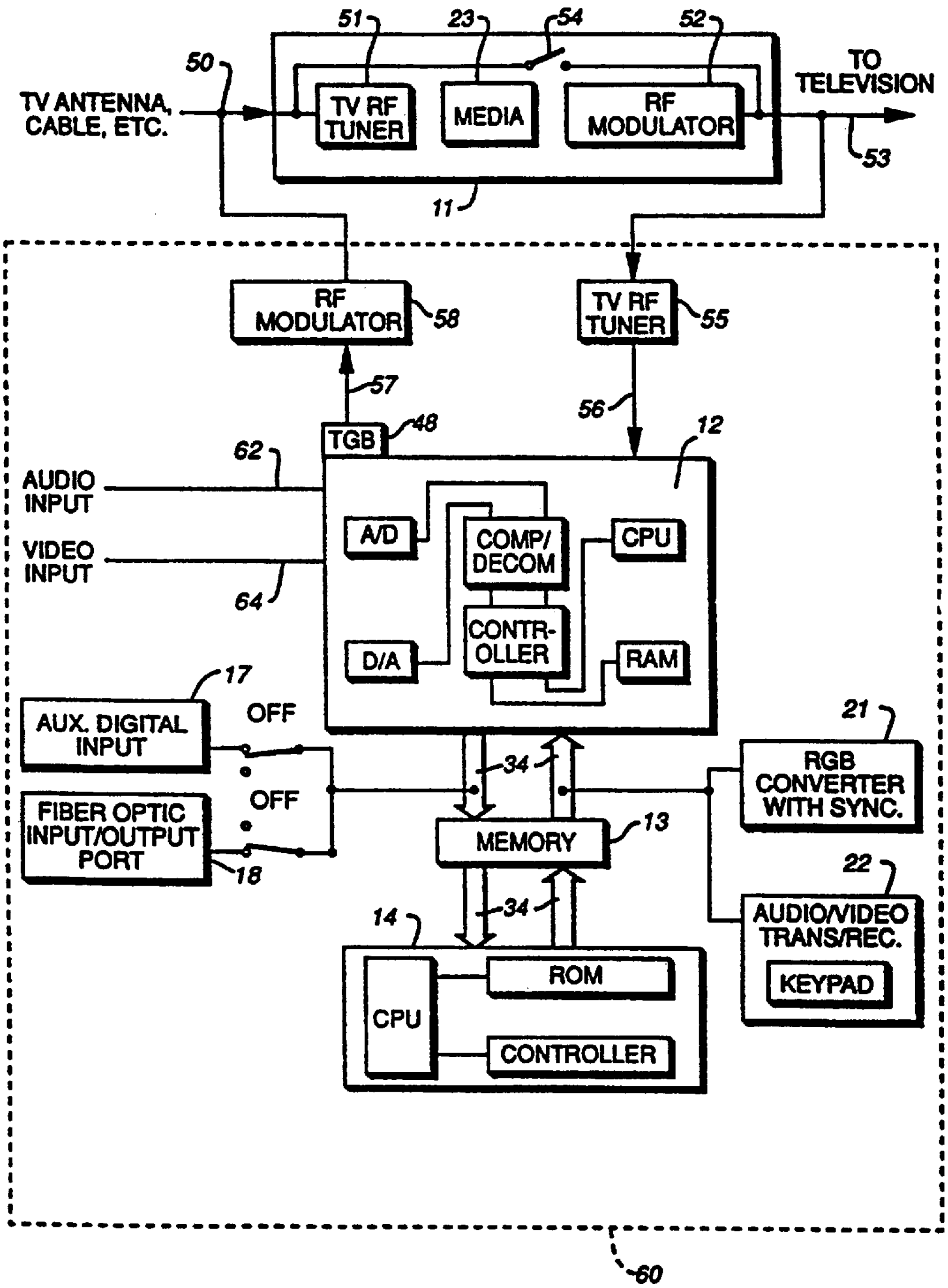
FIG._2



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TV ANTENNA, CABLE, ETC.
TV RF TUNER
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VIDEO INPUT
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A/D
COMP/ DECOM
CPU
D/A
CONTR-OLLER
RAM
17
AUX. DIGITAL INPUT
OFF
OFF
FIBER OPTIC INPUT/OUTPUT PORT
18
34
MEMORY
13
21
RGB CONVERTER WITH SYNC.
22
AUDIO/VIDEO TRANS/REC.
KEYPAD
14
34
CPU
ROM
CONTROLLER
60

FIG._3

BURST TRANSMISSION APPARATUS AND METHOD FOR AUDIO/VIDEO INFORMATION

RELATED APPLICATIONS

This is a continuation of application Ser. No. 08/624,958 filed on Mar. 28, 1996 abandoned, which is a continuation of Ser. No. 07/976,542 filed Nov. 16, 1992, abandoned, which is a division of Ser. No. 07/775,182 filed Oct. 11, 1991, U.S. Pat. No. 5,164,839, which is a continuation-in-part of Ser. No. 07/289,776 filed Dec. 27, 1988, U.S. Pat. No. 4,963,995.

BACKGROUND OF THE INVENTION

The video cassette recorder (VCR) has added significantly to the usefulness of the home television set. Important or exceptionally good programs may be recorded to be viewed again. Programs appearing at times that are inconvenient for viewing may be recorded for playback at a later time. Recorded movies or other materials, educational or entertaining, may be rented or borrowed for viewing at home. (As used in the remainder of this specification, the term "program" encompasses movies and other types of video and/or audio materials, whether broadcast from a TV station or another source.)

The typical VCR has its own tuner-receiver and a video-recorder. It can receive and record a program from one channel while the television set is being employed to view a program on another channel. Programs are recorded on magnetic tape. The tape is then played back and viewed on the television set. Features commonly included in the VCR are capabilities for advancing the tape forward or backward at a high speed, stopping motion at any frame to hold the image, or simply playing back the recording at normal speed.

Desirable features that are not normally available in a VCR are capabilities for copying recorded programs from one tape or alternative storage medium to a similar or dissimilar storage medium, editing recorded programs and high speed recording. Another desirable, but currently unavailable, feature is the capability for high speed, high quality transmission and reception by optical fiber using the VCR.

DESCRIPTION OF THE PRIOR ART

U.S. Pat. No. 4,768,110, incorporated herein by reference, describes a VCR having two tape decks included therein. The purpose for the inclusion of two decks rather than the usual single tape deck is to permit the simultaneous viewing of a live RF-modulated TV signal or prerecorded material while recording another live RF-modulated TV signal and to also allow the copying of material from a first magnetic cassette tape onto a second magnetic cassette tape without the use of a second VCR. Viewing of the recorded material during the copying process is also possible in this arrangement. A major disadvantage is that the incorporation of the second tape deck is expensive and limited to magnetic tape, and furthermore, this prior art does not allow for the transmission or reception of recorded material over optical fibers or the high speed reception or transmission of audio/video material in a digital format. An additional disadvantage is the inability for random access editing of the audio/video signal. Furthermore, the additional mechanical structure adds significantly to the overall dimension of the equipment and increases the prospects of mechanical failures.

SUMMARY OF THE INVENTION

In accordance with the invention, an improved audio/video recorder is provided with added features and functions which significantly enhance its usefulness and functionality.

It is, therefore, an object of the present invention to provide an improved audio/video recorder for use in conjunction with an ordinary home television set.

Another object of the invention is to provide in such an improved audio/video recorder a capability for transferring a previously recorded program from one magnetic tape or other storage medium to another.

A further object of the invention is to provide such a capability for transferring a recorded audio/video program without resort to the use of two magnetic tape decks, this being a cumbersome, limited, and expensive approach already proposed in the prior art.

A still further object of the invention is to provide an effective and efficient means for intermediate storage of the audio/video program in digital memory as a means for achieving the transfer of the audio/video program from one tape or storage medium to another.

A still further object of the invention is to provide in such an improved audio/video recorder a capability for accepting various forms of analog or digital audio and video input signals and for converting the analog input signals to digital form when appropriate.

A still further object of the invention is to provide in such an improved audio/video recorder a capability for editing the video input signals without the necessity of using multiple cassettes or recording media.

A still further object of the invention is to provide an improved audio/video recorder for connection to various signal sources including a TV RF tuner, video camera, video and audio line input, and direct audio/video digital input from sources as diverse as a fiber optic input line, a microwave transceiver or a computer.

A still further object of the invention is to provide an improved audio/video recorder having a capability for mixing live audio/video programs with either analog or digital audio/video input signals from another source

A still further object of the invention is to provide an improved audio/video recorder for simultaneously playing, viewing, recording and/or mixing digital and analog audio/video programs from different digital and analog audio/video sources or storage media.

A still further object of the invention is to provide an improved audio/video recorder which maximizes a given storage capacity, through the use of a data compression technique.

A still further object of the invention is to provide an audio/video recorder/transceiver utilizing a data compression technique for efficient storage of data, and efficient transmission and reception of a digitized audio/video program over a telephone line, a fiber optic cable, a microwave transceiver or other data transmission means.

A still further object of the invention is to provide in such an improved audio/video recorder a capability for delivering output signals in different forms or formats including a standard RF modulated output signal for viewing on a television set, a digital output signal for viewing on a high-resolution monitor, and audio output signals for a speaker system.

A still further object of this invention is to provide an improved audio/video recorder which provides for random

access to any given segment of a self-stored audio/video program so that the desired segment may be accessed and viewed without the time-consuming delays normally involved in fast-forward or fast-reverse searching procedures employed in present state-of-the-art VCR's.

A still further object of the invention is to provide an improved audio/video recorder which provides convenience in the editing of stored data by virtue of its random access memory capability.

A still further object of the invention is to provide an improved audio-video recorder which has the potential for enhanced audio and video quality by virtue of its capability for digital audio/video output and digital filtering techniques, and image or audio processing.

Further objects and advantages of the invention will become apparent as the following description proceeds, and the features of novelty which characterize the invention will be pointed out with particularity in the claims annexed to and forming a part of this specification.

BRIEF DESCRIPTION OF THE DRAWING

The present invention may be more readily described with reference to the accompanying drawing, in which:

FIG. **1** is a perspective view of the housing of the audio/video recorder editor/transceiver ("VCR-ET") disclosed and embodying the invention;

FIG. **1**A is an enlarged view of the circled area of FIG. **1**;

FIG. **2** is a functional block diagram of the VCR-ET of FIG. **1**;

FIG. **3** is a functional block diagram of a VCR-ET in accordance with another embodiment of the invention; and

FIG. **4** is a functional block diagram of an audio recorder/transceiver constructed in accordance with the invention.

DESCRIPTION OF THE PREFERRED EMBODIMENT

Referring to the drawing by reference characters, FIGS. **1** and **2** illustrate an improved audio/video recorder editor/transceiver **10** (VCR-ET) comprising an audio/video recording unit (AVRU) **11**, a video control unit (VCU) **12**, memory **13**, digital control unit (DCU) **14**, video line or camera input line **15**, TV RF tuner **16**, auxiliary digital input port **17**, fiber optic input/output port **18**, RF modulator **19**, RGB converter with synchronizer **21**, and an audio/video transmitter/receiver **22** with keypad **45**, all in a common housing.

The audio/video recording unit AVRU **11** may be a video cassette recorder similar to a conventional VCR in which the storage media **23** is a magnetic tape. Alternatively AVRU **11** may operate with other types of storage media including, but not limited to, other magnetic tape formats. AVRU **11** has all the functions of the typical VCR including record, play, rewind, slow motion, fast-forward and single frame hold.

An alternate form of storage media for use in AVRU **11** is the CD-ROM, which is a disk using a derivative of glass or plastic in conjunction with an aluminum or other metallic coating. Audio and video signals are stored in the form of irregularities in the aluminum coated surface and are read using a low power laser. In this case, the user would not be able to store or write on the CD-ROM, but would be able to play discs that have been recorded and distributed commercially. The storage of video and audio signals on the CD-ROM is in digital form which is readily accommodated by the video recorder of this invention.

Instead of using a CD-ROM, VCR-ET **10** can use optical discs as media **23**. Such optical discs are similar to a CD-ROM and use a variable power laser to read from or write on the disc.

A first type of optical disc may comprise a WORM (Write Once Read Many) optical disc. This device has the unique capability of writing on the disc permanently. A laser is used to change the magnetic or optical properties of the media. A lower-powered laser is then used to read the data from the disc. Data, in this case, is permanently recorded; it may neither be erased nor written over. A further description of this technology can be found in the November 1988 issue of *The Electronic System Design* magazine (ESD) pages 55–56, incorporated herein by reference.

A second and preferred type of optical disc to be used in AVRU **11** is an erasable optical disc. This disc has full read/write/erase capabilities. With this disc, AVRU **11** has the-same record/playback capabilities as a conventional VCR. As an example, erasable optical discs are used in Steven Jobs' "Next" machine as described in *Infoworld*, Volume 10, issue 42, pages 51 and 93, Oct. 17, 1988, incorporated herein by reference. In addition, the random access capabilities of the erasable disc (and of the CD-ROM and WORM) provide additional benefits as will be discussed in a later part of this specification.

A key element of VCR-ET **10**, which is responsible for its improved functionality, is the video control unit or VCU **12**. The VCU comprises an analog to digital converter (ADC) **24**, a digital to analog converter (DAC) **25**, a compressor/decompressor **26**, a controller **27**, a central processing unit (CPU) **28** and a random access memory (RAM) **29**. VCU **12**, using these elements, accomplishes the digitization and compression of analog signals as well as the reverse process in which the compressed digital signals are decompressed and converted back to analog signals.

As a first step in the processing of the composite video signals within VCU **12**, the sync signals are decoded to isolate signals for each picture frame for processing.

The video signals defining each frame may then be converted to a red analog signal, a green analog signal, and a blue analog signal in a conventional manner. The red, green and blue analog signals are then converted to digital form by the analog to digital converter (ADC) **24**. The frame is divided into a set of closely positioned rows and columns of picture elements or "pixels." Each pixel has a color defined by a set of three digital values defining strength of the primary color components, red, green and blue (RGB) respectively. In one embodiment, each frame is divided into an array of 300 by 300 pixels, with the color and luminance of each pixel being defined by a seven bit word for the red component, a seven bit word for the blue component, and a seven bit word for the green component. These words are generated by ADC **24**. The RGB video signal may also be processed by means of hue-saturation-intensity (HSI) color processing, where appropriate, as described in "Chips for Real-Time Comparisons," *Electronic Engineering Times*, issue 525, Feb. 13, 1989, page 122.

If each frame includes 90,000 pixels (300×300), and each pixel is defined by 21 bits (7 bits per primary color), the digital representation of a single video frame utilizes a sizable block of digital information (i.e., 1.89 megabits/frame) which must be processed very rapidly. (Approximately 30 frames/second are received from AVRU **11**.) Fortunately the analog to digital conversion of these signals may be accomplished at the desired speed using commercially available analog to digital converter integrated circuits. The analog to digital converter **24** (ADC) is a high-speed, high-accuracy, A to D "flash" converter avail-

able as a single IC (integrated circuit). Several different types of such A/D converters are available from Burr-Brown, one of which is the ADC 600. Part number TIC024, manufactured by Tektronix, Inc. is also appropriate. Other types of devices appropriate for this function are described in an article by K. Rogers entitled "8-bit A/D Flash Hits 500 Msamples", Electronic Engineering Times, Dec. 12, 1988, page 90, incorporated herein by reference.

Compression of the digital data defining a video frame and the reverse process (decompression) are accomplished by compressor/decompressor **26**. Various algorithms may be employed in the compression process which enable the representation of a series of numbers by a reduced number of digits. As an example, compression algorithms like CCITT Group IV may be used.

In one optional embodiment, to further reduce the amount of memory required to store a program, the compression algorithm can simply record data corresponding to only those pixels which change color from one frame to the next. This results in considerable memory space savings, since not all pixels change color each frame. Basing calculation upon 10% of the pixels changing from one frame to the next, it is estimated that memory requirements using this technique are cut by about 90%. It is also estimated that on the average, the CCITT Group IV algorithm can cut memory requirements by another 95%. Thus, if no data compression technique is used, it would take approximately 51.03 gigabytes to store a 2 hour video program, but by using the above compression techniques, it is estimated that memory **13** will require only 250 megabytes.

Controller **27** handles timing and aids in the communication between the different elements of VCU **12**, and between VCU **12**, AVRU **11** and memory **13**.

In one embodiment, the audio portion of the program is periodically sampled and digitized by analog to digital conversion. In one embodiment, this is done at a sample rate of 88,000/second, one byte per sample, to yield compact disc quality sound. The sampling rate could be dropped to reduce memory requirements. Also, the audio data can be compressed with conventional algorithms.

The process of converting either from analog to digital or from digital to analog requires memory for intermediate storage. Random Access Memory (RAM) **29** serves in this capacity. For this purpose either a DRAM (Dynamic RAM) or a SRAM (static RAM) may be employed. An example of a DRAM is the TI (Texas Instruments) TMX4C1024; an example of a SRAM is the INMOS IMS-1203. RAM **29** should have sufficient capacity to store at least two full uncompressed frames (e.g., about 472 KB).

The CPU (Central Processing Unit) **28** is a micro-**10** processor which controls the digitization process of VCU **12**. CPU **28** works with controller **27** to control and communicate with the other elements of the VCU. There are numerous commercially available microprocessors that are appropriate for this application. The Intel 80286, Intel 80386, Motorola 68020, and Motorola 68030 are examples. A more complete description of the microprocessors can be found in the Oct. 27, 1988 issue of *Electronic Design News* (EDN), pages 231 and 242, incorporated herein by reference, or in the applicable data sheets.

Controller **27**, CPU **28** and RAM **29** serve in the same manner during the reverse processes, i.e., decompression and digital to analog conversion. Decompression is first accomplished in compressor/decompressor **26**. The decompressed digital signal is then converted to an analog signal by digital to analog converter (DAC) **24** (assuming its destination requires an analog form). In the course of converting the decompressed signals from the VCU **12** for use by the AVRU **11** the signals are synchronized by the time base generator (TBG) or corrector **48**. TBG generator **48** inserts synchronization pulses into the signal provided by VCU **12** to identify individual raster scan lines and frames so that the resulting signal can be used by a conventional television set or VCR. TBG **48** can be bypassed by shunt switch **48'** for the purpose of transmitting either compressed or decompressed signals from VCU **12** directly to the AVRU **11** in an uncorrected time based mode.

DAC **25** provides the inverse of the function performed by A/D converter **24**. DAC **25** is a high-speed, high accuracy digital to analog converter. An example of such a converter is the Burr-Brown DAC60 digital to analog converter.

Different types of memory technologies are adaptable for use in memory **13**. As mentioned earlier, DRAM and SRAM semiconductor memories are commonly used for applications of this type and are readily available.

One type of random access memory is CMOS (Complimentary Metal Oxide Semiconductor). The CMOS memory has the advantage of a relatively low power requirement and is readily adaptable for use of battery backup for semi-permanent data storage. Other types of memory include the above mentioned optical disc memories, bubble memories and magnetic disks. Another appropriate data storage media may be "Digital Paper" available from ICI Image data of Wilmington, Del.

Emerging memory technologies may also prove advantageous with capabilities for mass data storage in even smaller physical dimensions.

Digital Control Unit (DCU) **14** comprises a CPU (Central Processor Unit) **31**, a ROM (Read Only Memory) **32** and a controller **32**. DCU **14** is responsible for all of the digital editing processes. Through the use of DCU **14**, video segments may be edited and rearranged. Thus, one may use DCU **14** to rearrange the scenes in a program, alter the program sound track, etc.

In addition, a program may be edited, one frame at a time, by changing the contrast, brightness, sharpness, colors, etc. (Alteration of the contrast, brightness, sharpness and colors can be automated as well.) In one embodiment, images can be rotated, scaled (i.e., made larger or smaller), etc. In addition, pixel by pixel editing can be accomplished by DCU **14**, e.g., in a manner similar to a PC paint program. Similar editing features can be incorporated for the audio portion of each program. In one embodiment, a display such as a flat panel video display (not shown) is built into the VCR-ET. A user interface control panel of DCU **14** allows a user to select a desired frame number from a menu on the display. The VCR-ET then displays a strip of frames (including several frames before and after the selected frame). The user can delete frames in a strip, select a point where other frames are to be inserted into the program, or edit different frames (i.e., alter contrast, brightness, sharpness, colors, etc.). In one embodiment, a user input device such as a light pen or mouse can be used to select individual frames in a strip for editing.

Instead of incorporating a flat display into VCR-ET **10**, in another embodiment, a television coupled to output lead **42** of RF modulator **19** can be used during editing.

CPU **31** is a microprocessor of the type described in connection with the CPU **28** of VCU **12**. Controller **33** is an integrated circuit which handles the timing and interfacing between DCU **14** and memory **13**. ROM **32** holds the necessary step-by-step editing programs which are installed

at the factory. A currently available example of a suitable ROM for this application is the Texas Instruments part TMS47256. CPU **31** and controller **33** together control the editing process as they execute the programs stored in ROM **32**.

The VCU **12**, memory **13** and DCU **14** communicate with each other via a high speed data bus **34**. The high speed data bus is required in order to meet bandwidth requirements. Examples of suitable data bus devices are Motorola's VME bus, Intel's Multibus and the Optobuss (U.S. Pat. No. 4,732,446).

A video line or camera input line **15** is provided to enable VCR-ET **10** to receive an input signal from a source such as a television camera, a conventional VCR, a television tuner, or another VCR, etc. The signals received at input line **15** are typically carried by a coaxial cable and are in the form of a standard television composite signal. As used throughout this specification, the words "standard television composite signal" or its acronym STCS shall be read to include any one of the following: NTSC, PAL, SECAM, HDTV, or any American or European broadcast signal standards. (NTSC, PAL and SECAM are discussed in "Reference Data for Radio Engineers", published by Howard W. Sams & Co. in 1983, incorporated herein by reference.) An NTSC composite signal is defined as the analog signal that carries the chrominance (color), luminance (brightness), synchronization (timing) and audio signals that make up the video signals received and displayed by television and video cassette recorders. These four components are combined into one signal by modulating the components in different ways. (Amplitude modulation and phase modulation are examples.) The standard video line signal is such a composite signal and may be received at input line **15** from one of the above-mentioned sources.

TV RF tuner input port **16** also supplies a composite signal as described in regard to video input line **15**. The difference is that this signal is received from an antenna or cable TV coaxial cable. To receive such a signal, tuner **16** is capable of being set or tuned to receive the desired carrier frequency or television channel.

Selector switch **35** is provided to select either video input line **15** or TV RF tuner **16** as an input signal source to AVRU **11**.

Auxiliary digital input port **17** is employed to receive any acceptable digital signal such as computer-generated video signal or as may be supplied by another VCR-ET. This signal, for example, may be an RGB video signal such as that delivered to computer monitors, or it may be a digitized audio signal. (As mentioned above, an RGB signal is a signal which communicates the strength of the red, green and blue color components for the pixels that make up each video frame.) Switch **36** selects whether the digital video/audio input signal is chosen from auxiliary digital input port **17**. Switch **36** supplies the selected signal to high speed data bus **34** which carries the signals in digital form.

Fiber optic port **18** incorporates a fiber optic transceiver. Port **18** has a capability for transforming fiber optic (light) signals to electrical signals or for transforming electrical signals to fiber optic signals. Port **18** thus provides a capability for two-way communication between high speed data bus **34** and a fiber optic signal line. The incorporation of fiber optic port **18** in the VCR-ET provides a capability for receiving audio/video signals from or delivering audio/video signals to the fiber optic line such as a fiber optic telephone line. The fiber optic line carries digital signals in the form of light waves over great distances with a high

degree of accuracy and reliability and at a high speed (e.g., about 200 megabytes/second). The VCR-ET can receive/transmit a video program at an accelerated rate via fiber optic port **18** from/to a variety of sources. For example a video program may be communicated at an accelerated rate from the first VCR-ET to a second VCR-ET in less time than it would take to view the program. Thus, it is not necessary to access the optical fiber for long periods of time in order to transmit a long video program.

It is also envisioned that in the future, a video library may be established which downloads video programs at an accelerated rate via optical fibers to a subscriber's VCR-ET. After downloading, the program may be viewed, stored in memory, edited and/or a hard copy of the program may be made on magnetic tape, optical disk, etc.

Switch **37** is provided to select connection to the fiber optic input/output port **18**. An OFF or open position is provided. The selected signal is delivered to or supplied from high speed data bus **34**.

Analog output signals from AVRU **11** are delivered to the common terminal **38** of a selector switch **39**. When set to position A, switch **39** delivers the output signal of AVRU **11** directly to a video output line **41** as a standard STCS composite signal; when set to position B switch **39** delivers the output of VRU **11** to the input of RF modulator **19**. Modulator **19** converts the video signal to an RF-modulated composite signal for delivery to such devices as televisions and conventional VCR's. These types of devices play back the video program on a particular frequency channel (such as channel 4) on the television. Delivery to the television or VCR is via RF output line **42**.

Digital output signals from VCR-ET **10** may be dispatched from high speed data bus **34** via line **43** to input leads of RGB converter **21** and audio/video transmitter/receiver **22**.

RGB converter **21** converts the STCS signal into an RGB signal as required by computer monitors and similar display devices. The converted signal is received by a display device connected to RGB converter output line **44**.

VCR-ET **10** includes audio/video transmitter/receiver **22** which is typically a built-in modem. Advantageously, the modem may be used to communicate an audio/video program over conventional phone lines in a manner similar to that described above with respect to optical fibers. The term modem is derived directly from its functionality as a modulator-demodulator which allows transfer of the audio/video signal in a digital format over the standard telephone line. Modems are commonly available for computers and are currently available in the form of a single integrated circuit. As an example, Sierra Semiconductor offers a 2400 baud single chip modem under its part number SC111006. Representative manufacturers of these single modem IC's can be found in the Apr. 14, 1988 issue of Engineering Design News (EDN), pages 124–125. Some of these single IC modems have the added capability of generating the tones for dialing a phone number. The destination phone number may be entered by means of an optional keyboard/keypad **45** incorporated in the video recorder **10** of the invention. Output port **46** of transmitter/receiver **22** connects directly to the telephone line.

Also associated with Modem **22** is an auxilliary keyboard **45'** (FIG. **1**A) of buttons for commanding the modem to perform tasks such as starting a transmission over phone lines (**45***a*), terminating a transmission (**45***b*), automatic telephone answering to receive transmissions (**45***c*), using an optional speaker (not shown) to monitor phone lines (**45***d*),

using an optional microphone (not shown) to speak over the phone lines (**45***e*) and for controlling the baud rate (**45***f*).

The application and utilization of the VCR-ET may include a number of forms or operating modes. In its first and simplest operating mode, AVRU **11** may be operated in the manner of a conventional VCR with signals from an antenna being received by tuner **16** and recorded directly on media **23** in analog form. At the same time the received program may be viewed on the television screen with the television connected at video output terminal **42**. An optional signal source for this type of operation is the video line or camera input line **15** selectable by switch **35**.

In a second operating mode a program stored in media **23** of AVRU **11** may be played back and viewed on the connected television set.

When it is desired to copy a program from one recording media to another, the recording media holding the desired program is installed in the AVRU. The recording media is then played back with optional viewing on a connected television set or other TV monitor or listening through speakers (as appropriate). As the recording media is played back, the analog signals from the recording media (video and/or audio) are dispatched to VCU **12** via connection **47**. The analog signals are converted to digital signals by ADC **24**, compressed by compressor/decompressor **26** and the compressed digital signals are stored in memory **13**. The foregoing operations are accomplished under the control of controller **27** and CPU **28**. RAM **29** is used for interim data storage during this process. Once the complete video/audio program has been stored in memory **13**, the recording media from which the stored program has just been read is replaced by blank recording media upon which the stored program is to be copied. CPU **28** in cooperation with controller **27** and RAM **29** then executes the decompression and digital to analog conversion of the program stored in memory **13**, decompression taking place in compressor/decompressor **26**, and digital to analog conversion being accomplished by DAC **25**. The resulting analog program is stored on the blank recording media which constitutes media **23** of AVRU **11**.

In an alternate mode of operation, the decompression circuitry of VCU **12** can be bypassed. Thus, a user has the option of downloading the stored program from memory **13** onto recording media **23** in compressed digital format. The user can then reload the program from media **23** into memory **13** at a future time for viewing, editing or recording back onto recording media **23** in analog form. This capability allows the user to quickly clear memory **13** for other interim uses and also provides the user with a hard copy of the program in digital format. The hard copy in compressed digital format has a number of uses, e.g. it could be archived for later viewing, transmitted by an appropriate independent transmitter, etc.

During the foregoing procedures, DCU **14** may be utilized for editing operations. As the program is being read from the first or original recording media, it is simultaneously viewed on the TV screen, or listened to by means of an audio monitor, converted to digital signals, compressed and stored in memory **13**. Once the digital audio/video program is stored in memory **13**, editing is accomplished by the user through control of DCU **14**, by means of a control panel (not shown) coupled to DCU **14**. If desired, additional audio/video signals may be simultaneously entered into memory **13** and added to those received from VCU **12**. The additional signals may be introduced from auxiliary digital input port **17** or from fiber optic input/output port **18** and may comprise video captions for super imposed position upon the stored video images, or they may be audio commentaries to be added to silent video presentations. In addition, as mentioned above, the order in which various segments appear in the video programs may be altered. Certain undesired segments, such as TV commercials, may be removed. This editing operation is accomplished under the control of DCU **14**.

In still another operating mode, a program stored in media **23** of AVRU **11** or being received by AVRU **11** from input line **15** (as from a video camera) may be digitized and compressed by VCU-**12** and routed via bus **34**, to memory **13**. The data from memory **13** is then routed to line **43**, transmitter/ receiver **22** and to a telephone line. At the other end of the telephone line the signals received are processed by another VCR-ET.

Once received in the second VCR-ET's memory **13**, the digitized program can then either be viewed directly from memory or transferred to storage medium **23**, either in its entirety or in random segments, based on user preference.

In the case of video camera input at input **15** the transmitted signals may comprise a live transmission. Alternatively the transmitted program may be derived from a program stored in media **23** of AVRU **11**. In this case the stored analog program is again decoded, digitized, compressed and transmitted via bus **34** to memory **13**. The data in memory **13** is then communicated via line **43** and transmitter/receiver **22** to telephone lines.

It follows, of course, that digitized video and audio signals from the remote VCR-ET at the other end of the telephone line may be received at line **46**, entered into memory **13** via transmitter/receiver **22**, converted to analog signals by VCU **12**, and recorded on media **23** and then viewed, if desired, on a television set connected at output **42**. A hard copy of the program may also be made for later viewing.

As mentioned earlier, when any of the foregoing operations entail the processing of unmodulated video signals, such signals must first be processed by RF modulator **19** before they can be accepted by devices such as a conventional VCR or television set; when the monitoring means is a computer monitor or a similar display device the signals are processed by RGB converter **21**.

All of the foregoing operations are performed with enhanded quality and efficiency by virtue of the digital, rather than analog, storage and transmission modes and the compressed data storage mechanism, with additional advantages of improved cost and reliability afforded in the case of tape to tape (or other media to media) program transfers by virtue of the requirement for only a single tape deck or other storage device.

FIG. **3** illustrates an alternative embodiment invention in which AVRU **11** is not integral with VCU **12**, memory **13** or editor **14**. In this embodiment, AVRU **11** is a conventional, commercially available VCR which receives a modulated video input signal on an input cable **50**. In this embodiment AVRU **11** includes a RF tuner **51** for demodulating the input signal so it can be stored in media **23**. AVRU **11** also includes a RF modulator **52** for modulating the signal received from media **23** and providing the RF modulated output signal on an output cable **53**, which can be coupled to a television set. (TV RF tuner **51** and RF modulator **52** are provided in typical commercially available VCR's.) A switch **54** is provided to couple input cable **50** to output cable **53** when media **23** is not serving as a video signal source. The VCR-ET of this embodiment includes a TV RF tuner **55** which receives and demodulates the signal on cable

53, and provides the resultant analog audio/video signal on a lead **56**, which is digitized and compressed as described above. In this alternative embodiment, the digitized compressed signal may be processed as described above, e.g. stored in memory **13** (via high speed bus **34**), edited, transmitted by the fiber optic port **18** to another VCR-ET, etc. When it is desired to view a program stored in memory **13**, data from memory **13** is decompressed and converted to an analog signal by VCU **12**, and the resulting signal is provided on an output lead **57** to a RF modulator **58**, which modulates the video signal so that it can be received and stored by AVRU **11** or viewed on a television coupled to cable **53**. (As mentioned above, in the FIG. **3** embodiment, AVRU **11** is a conventional VCR.)

One advantage of the embodiment of FIG. **3** is that many people already own VCR's. Rather than buying apparatus which duplicates much of the hardware already present in their VCR, the embodiment of FIG. **3** would provide to owners of conventional VCR's capabilities which are otherwise currently unavailable in an economical manner.

In one embodiment, analog auxiliary audio and video input terminals **62**, **64** are provided so that analog signals may be provided by alternate sources to VCU **12**.

The embodiments described above include means for transmitting/receiving video programs over fiber optic cables. However, in an alternative embodiment, either in place of fiber optic port **18** or in addition to fiber optic port **18**, means are provided for transmitting and/or receiving a video program via microwave. In conventional microwave technology, satellite systems and microwave transmitters transmit data using a low power/high frequency signal. In an embodiment of the invention designed to receive microwaves, the VCR-ET includes an amplifier for amplifying the microwave signal and a demodulator for obtaining the video program signal from the microwave signal. Receiving, amplifying and demodulating the microwave signal can be accomplished with conventional microwave transceiver equipment. The video program signal is typically in digital form, and may be stored, viewed or edited as in the above-described embodiments. Program data from memory **13** can also be transmitted by the microwave transciever, thereby providing the capability for microwave transmission of stored video programs in compressed digital format. Thus, the invention can be used to receive and transmit programs via microwaves at an accelerated rate similar to and at least as fast as, the transmission and reception of programs over optical fibers. This feature allows transmission and reception of programs in a few minutes or seconds using currently available technology. Both point-to-point microwave transceivers and satellite transceivers may be used.

The embodiments described include means for receiving, storing and transmitting both audio and video signals. However, the invention encompasses apparatus which can store and transmit video signals only and apparatus which can store and transmit audio signals only. An embodiment designed to store and compress audio signals is illustrated in FIG. **4**. Referring to FIG. **4**, an audio signal source **70** (a tape recorder, microphone, record player, etc.) is coupled to a digitizer and compressor circuit **72**, which converts the analog signal to a digital signal and compresses the digital signal in a manner similar to VCU **12** described above. The digital compressed signal can then be stored in a memory **74**. Of importance, data from memory **74** can be transmitted by a fiber optic transceiver **76**, or by a microwave transceiver **78** at an accelerated rate. This is important not only in a home entertainment application, but in other applications as

well. For example, a user can dictate an audio presentation and send it to a remote location (e.g. an office) at an accelerated rate without having to monopolize the transmission medium (e.g. the fiber optic cable) for an extended length of time.

The business uses of the embodiment illustrated in FIG. **4** makes home offices feasible for many workers now confined to more traditional offices and also opens new possibilities to business people who are traveling.

In the embodiment of FIG. **4**, data can also be loaded from memory **74**, via a modem **79** over a conventional phone line **80**. Data can also be received from phone line **80**, fiber optic transceiver **76** or microwave transceiver **78**, loaded into memory **74**, and converted to an analog signal by circuit **72**, to be listened to via an audio monitor **82**, or to be stored on an audio tape cassette **84** or other storage media.

An editor **86** is optionally provided so that the data in memory **74** may be edited, e.g., by rearranging the order of portions of the audio program, increasing or decreasing the volume of portions (or different frequency components) of audio program, or enhancing the audio program through filtering techniques (e.g. to remove static and noise).

An improved audio/video recorder with significantly expanded functional capabilities is thus provided in accordance with the stated objects of the invention and although but a single embodiment of the invention has been illustrated and described, it will be apparent to those skilled in the art that various changes and modifications may be made therein without departing from the spirit of the invention or from the scope of the appended claim. For example, the VCR-ET can be constructed so as to be portable. Thus, it could be carried to a location where it is desired to record a program, and used to edit the program after it is recorded with a video camera. Other modifications will be apparent to those skilled in the art in light of the present specification.

What is claimed is:

1. An audio/video transceiver apparatus comprising:

input means for receiving audio/video source information, said audio/video source information comprising a multiplicity of video frames collectively representing at least one full motion video program;

compression means, coupled to said input means, for compressing said audio/video source information into a digital time compressed representation thereof, wherein said digital time compressed representation of said audio/video source information is capable of being transmitted in a burst transmission time period that is substantially shorter than a time period associated with real time viewing by a receiver of said audio/video source information;

storage means, coupled to said compression means, for storing said digital time compressed representation of said audio/video source information; and

transmission means, coupled to said storage means, for transmitting said digital time compressed representation of said audio/video source information away from said audio/video transceiver apparatus in said burst transmission time period.

2. The audio/video transceiver apparatus of claim **1**, further comprising editing means, coupled to said storage means, for editing the digital time compressed representation of said audio/video source information stored in said storage means and for storing the edited digital time compressed representation of said audio/video source information in said storage means.

3. The audio/video transceiver apparatus of claim **2**, wherein said transmission means is configured to receive the

edited digital time compressed representation of said audio/video source information and to transmit the edited digital time compressed representation of said audio/video source information away from said audio/video transceiver apparatus in said burst transmission time period.

4. The audio/video transceiver apparatus of claim **1**, further comprising:

decompression means, coupled to said storage means, for selectively decompressing the digital time compressed representation of said audio/video source information stored in said storage means; and

editing means, coupled to said decompression means and said storage means, for editing the decompressed digital time compressed representation of said audio/video source information, and for then storing the edited decompressed digital time compressed representation of said audio/video source information in said storage means.

5. The audio/video transceiver apparatus of claim **1**, wherein said input means comprise analog to digital converter means for converting analog audio/video source information received at said input means to corresponding digital audio/video source information.

6. An audio/video information transfer network comprising a plurality of audio/video transceivers coupled via at least one communication link, each of the audio/video transceivers comprising:

input means for receiving audio/video source information, said audio/video source information comprising a multiplicity of video frames collectively representing at least one full motion video program;

compression means, coupled to said input means, for compressing said audio/video source information into a digital time compressed representation thereof, wherein said digital time compressed representation of said audio/video source information is capable of being transmitted in a burst transmission time period that is substantially shorter than a time period associated with real time viewing by a receiver of said audio/video source information;

storage means, coupled to said compression means, for storing said digital time compressed representation of said audio/video source information; and

transmission means, coupled to said storage means, for transmitting said digital time compressed representation of said audio/video source information away from said audio/video transceiver apparatus in said burst transmission time period.

7. The audio/video transfer network of claim **6**, wherein:

said input means of at least one of said plurality of audio/video transceivers includes a fiber optic input port;

said transmission means of at least one other of said plurality of audio/video transceivers includes a fiber optic output port; and

said at least one communication link includes a fiber optic transmission line coupling in communication said fiber optic input port with said fiber optic output port.

8. The audio/video transfer network of claim **6**, wherein said transmission means of at least one of said plurality of audio/video transceivers includes a modem, and said at least one communication link includes a telephone transmission line.

9. The audio/video transfer network of claim **6**, wherein at least one of said audio/video transceivers further comprises editing means, coupled to said storage means, for editing the digital time compressed representation of said audio/video source information stored in said storage means and for storing the edited digital time compressed representation of said audio/video source information in said storage means.

10. The audio/video transfer network of claim **6**, wherein at least one of said audio/video transceivers further comprises:

decompression means, coupled to said storage means, for selectively decompressing the digital time compressed representation of said audio/video source information stored in said storage means; and

editing means, coupled to said decompression means and said storage means, for editing the decompressed digital time compressed representation of said audio/video source information, and for then storing the edited decompressed digital time compressed representation of said audio/video source information in said storage means.

11. The audio/video transceiver network of claim **6**, wherein at least one of said plurality of audio/video transceivers further comprises analog to digital converter means for converting analog audio/video source information received at said input means to corresponding digital audio/video source information.

12. A method for handling audio/video source information, the method comprising the steps of:

receiving audio/video source information, said audio/video source information comprising a multiplicity of video frames collectively constituting at least one full motion video program;

compressing the received audio/video source information into a digital time compressed representation thereof, the digital time compressed representation of said audio/video source information having an associated burst transmission time period that is substantially shorter than a time period associated with real time viewing by a receiver of said audio/video source information;

storing the digital time compressed representation of said audio/video source information; and

transmitting, in said burst transmission time period, the stored digital time compressed representation of said audio/video source information to a selected destination.

13. The method of claim **12**, further comprising the steps of:

editing the stored time compressed representation of said audio/video source information; and

storing the edited time compressed representation of said audio/video source information.

14. The method of claim **12**, further comprising the step of converting the received audio/video information from an analog format to a digital format.

15. The method of claim **12** wherein the step of transmitting the stored digital time compressed video information further comprises sending said time compressed data to one of a plurality of audio/video transceivers connected over at least one communications link.

16. The method of claim **15** wherein said at least one communications link comprises an optical channel.

17. The method of claim **15**, wherein said at least one communications link comprises a telephone transmission channel.

18. The method of claim **12**, further comprising the step of providing a network that includes a plurality of audio/

video transceivers, coupled via at least one communications link, said selected destination comprising at least one of said plurality of audio/video transceivers.

19. The method of claim **18**, wherein said at least one communications link comprises an optical channel.

20. The method of claim **18**, wherein said at least one communications link comprises a telephone transmission channel.

21. A method for handling audio/video source information, the method comprising the steps of:

receiving audio/video source information as a digital time compressed representation thereof, said audio/video source information comprising a multiplicity of video frames collectively constituting at least one full motion video program selected from a video library storing a plurality of video programs in a digital time compressed representation thereof for selective retrieval;

said at least one video program being received by a receiver in a burst transmission time period that is substantially shorter than a time period associated with real-time viewing by a receiver of said at least one video program;

storing the digital time compressed representation of said audio/video source information; and

transmitting, in said burst transmission time period, the stored digital time compressed representation of said audio/video source information to a selected destination.

22. The method of claim **21**, further comprising the step of providing a network that includes a plurality of audio/video transceivers, coupled via at least one communications link, said selected destination comprising at least one of said plurality of audio/video transceivers.

23. The method of claim **22**, wherein said at least one communications link comprises an optical channel.

24. The method of claim **22**, wherein said at least one communications link comprises a telephone transmission channel.

* * * * *

UNITED STATES PATENT AND TRADEMARK OFFICE

CERTIFICATE OF CORRECTION

PATENT NO. : 5,995,705
DATED : November 30, 1999
INVENTOR(S) : Richard Lang

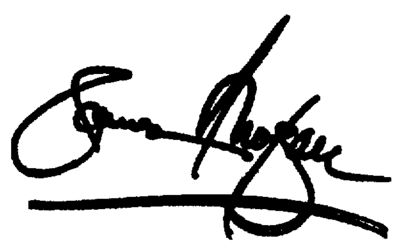
It is certified that error appears in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

Title page,
Related U.S. Application Data, should read as follows:
-- [63] Continuation of application No. 08/624,958, filed March 28, 1996, now abandoned, which is a continuation of application No. 07/976,542, filed Nov. 16, 1992, now abandoned, which is a division of application No. 07/775,182, filed November 11, 1991, now Pat. No. 5,164,839, which is a division of application No. 07/347,629, filed May 5, 1989, now Pat. No. 5,057,932, which is a continuation-in-part of application No. 07/289,776, filed December 27, 1988, now Pat. No. 4,963,995.

Signed and Sealed this

Thirtieth Day of April, 2002

Attest:



JAMES E. ROGAN

Attesting Officer

Director of the United States Patent and Trademark Office

EXHIBIT D

United States Patent [19]

Lang

[11] **Patent Number:** **5,057,932**

[45] **Date of Patent:** **Oct. 15, 1991**

[54] **AUDIO/VIDEO TRANSCEIVER APPARATUS INCLUDING COMPRESSION MEANS, RANDOM ACCESS STORAGE MEANS, AND MICROWAVE TRANSCEIVER MEANS**

[75] Inventor: **Richard A. Lang**, Cave Creek, Ariz.

[73] Assignee: **Explore Technology, Inc.**, Scottsdale, Ariz.

[21] Appl. No.: **347,629**

[22] Filed: **May 5, 1989**

Related U.S. Application Data

[63] Continuation-in-part of Ser. No. 289,776, Dec. 27, 1988, Pat. No. 4,963,995.

[51] **Int. Cl.**[5] **H04N 5/76**

[52] **U.S. Cl.** **358/335**; 358/133; 360/8; 360/9.1; 360/14.1

[58] **Field of Search** 381/29, 31, 32, 34, 381/35; 375/122; 370/109; 360/9.1, 32, 8; 358/133, 134, 135, 136, 137, 138, 261.1, 261.2, 261.3, 465; 379/100

[56] **References Cited**

U.S. PATENT DOCUMENTS

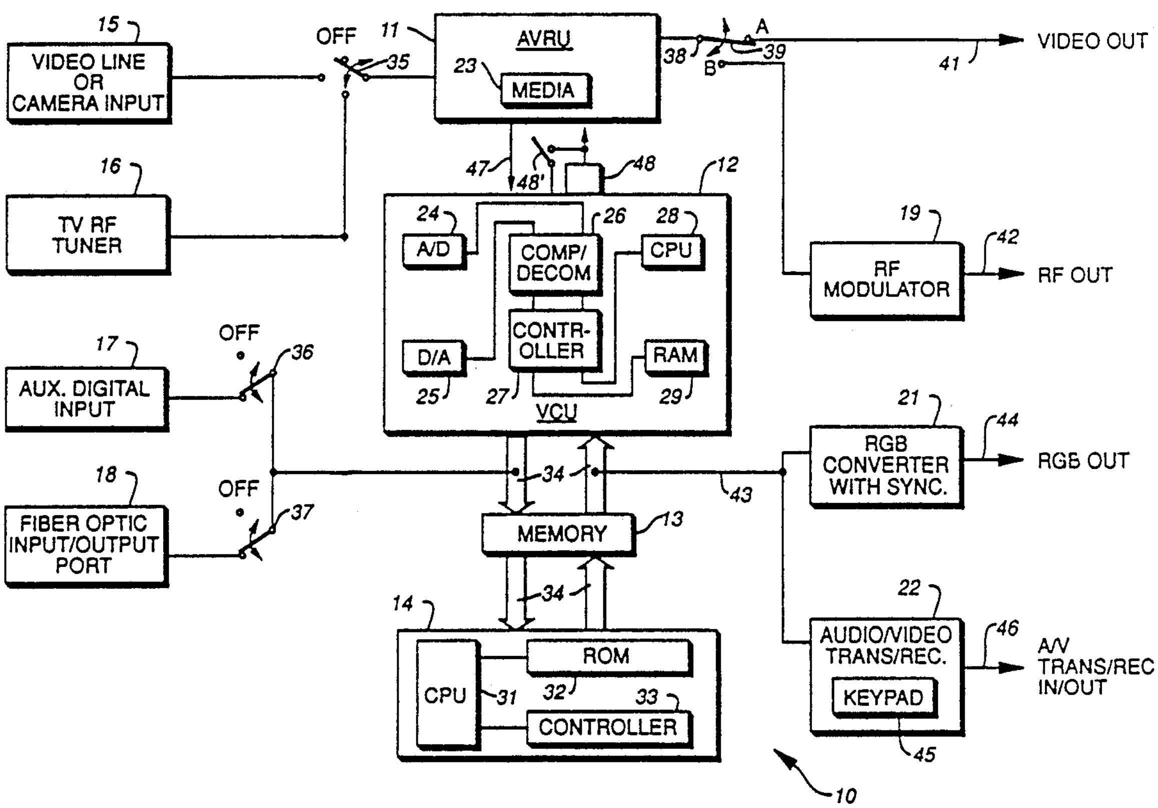
4,179,709	12/1979	Workman	358/133
4,400,717	8/1983	Southworth et al.	358/134
4,516,156	5/1985	Fabris et al.	379/53
4,698,664	10/1987	Nichols et al.	358/311
4,709,418	11/1987	Fox et al.	358/86
4,724,491	2/1988	Lambert	358/310
4,768,110	8/1988	Dunlap et al.	360/61
4,774,574	9/1988	Daly et al.	358/133
4,851,931	7/1989	Parker et al.	360/15

Primary Examiner—Roy N. Envall, Jr.
Assistant Examiner—Huy Nguyen
Attorney, Agent, or Firm—William E. Hein

[57] **ABSTRACT**

An improved video recorder/transceiver with expanded functionality ("VCR-ET") including a capability for storing video and video programs in digital format, editing such programs, transferring such programs onto a hard copy magnetic media, and transmitting such programs to a remote location using a second VCR-ET. The increased functionality is realized through the use of analog to digital conversion, signal compression and intermediate storage in an integrated circuit, random access memory. The recorder/transmitter has capabilities to transmit and receive program information in either a compressed or decompressed format over fiber optic lines, conventional phone lines or microwaves.

5 Claims, 4 Drawing Sheets



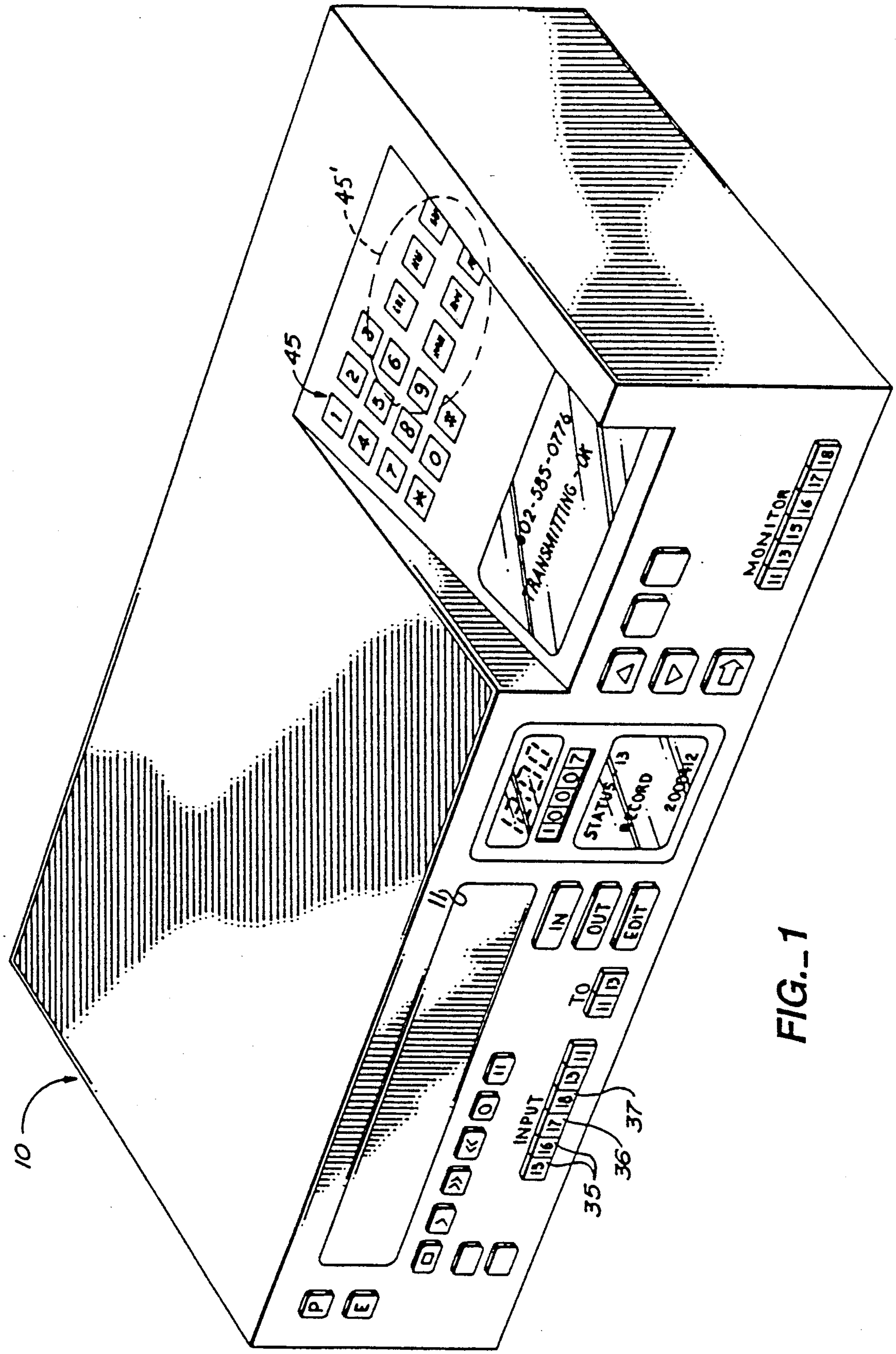

10
11
45
45'
35
36
37
INPUT
MONITOR
IN
OUT
EDIT
TO
STATUS
RECORD
TRANSMITTING ~OK

FIG._1

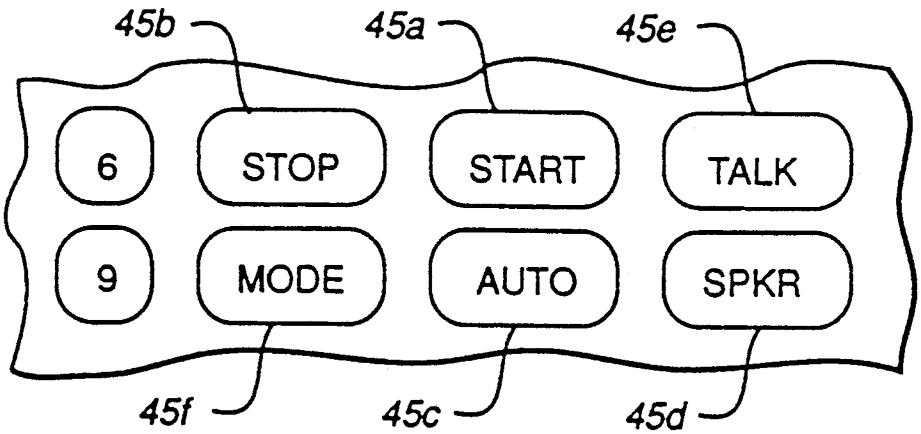

45b
45a
45e
6
STOP
START
TALK
9
MODE
AUTO
SPKR
45f
45c
45d

FIG._1A

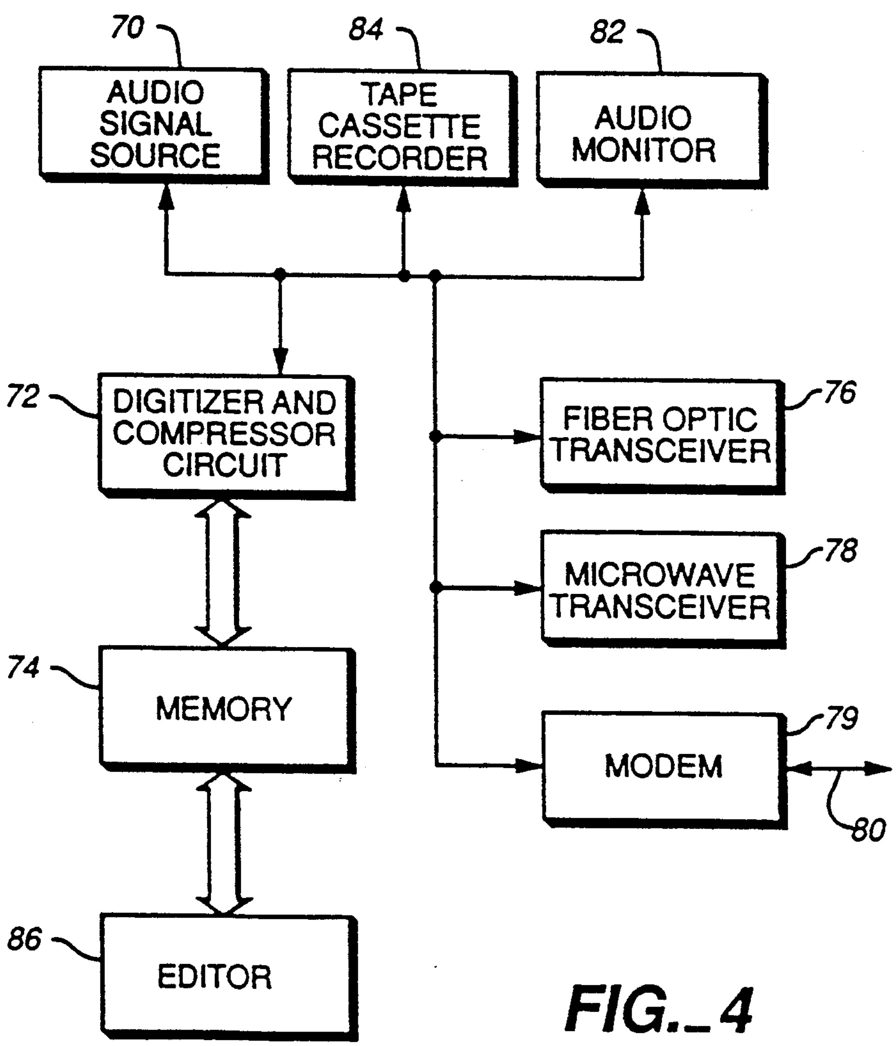
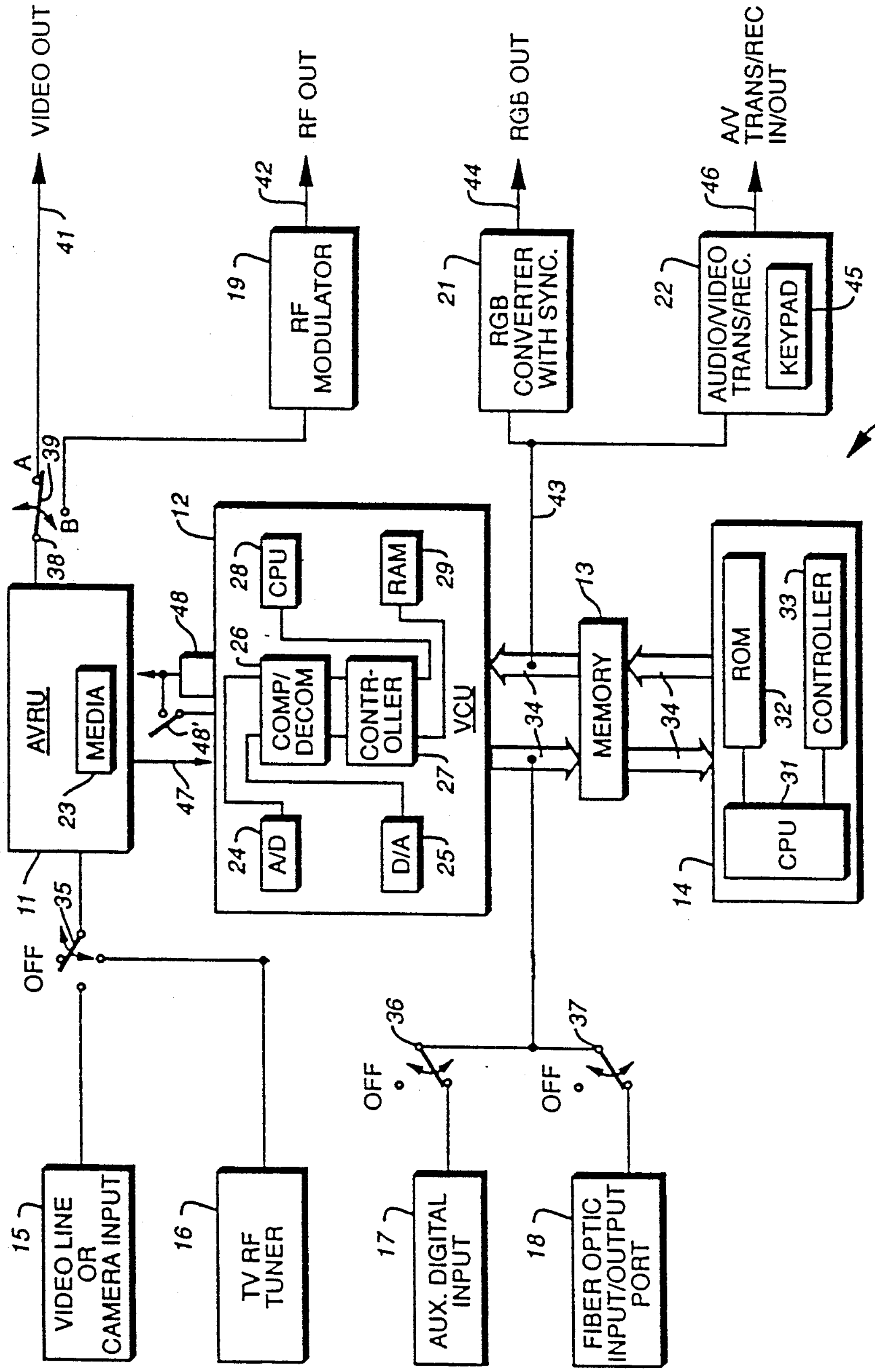

70
84
82
AUDIO SIGNAL SOURCE
TAPE CASSETTE RECORDER
AUDIO MONITOR
72
DIGITIZER AND COMPRESSOR CIRCUIT
FIBER OPTIC TRANSCEIVER
76
MICROWAVE TRANSCEIVER
78
74
MEMORY
MODEM
79
80
86
EDITOR

FIG._4


VIDEO OUT
RF OUT
RGB OUT
A/V TRANS/REC IN/OUT
RF MODULATOR
RGB CONVERTER WITH SYNC.
AUDIO/VIDEO TRANS/REC.
KEYPAD
AVRU
MEDIA
CPU
RAM
COMP/ DECOM
CONTR-OLLER
VCU
A/D
D/A
MEMORY
ROM
CONTROLLER
CPU
OFF
A
B
VIDEO LINE OR CAMERA INPUT
TV RF TUNER
AUX. DIGITAL INPUT
FIBER OPTIC INPUT/OUTPUT PORT
OFF
OFF
10
11
12
13
14
15
16
17
18
19
21
22
23
24
25
26
27
28
29
31
32
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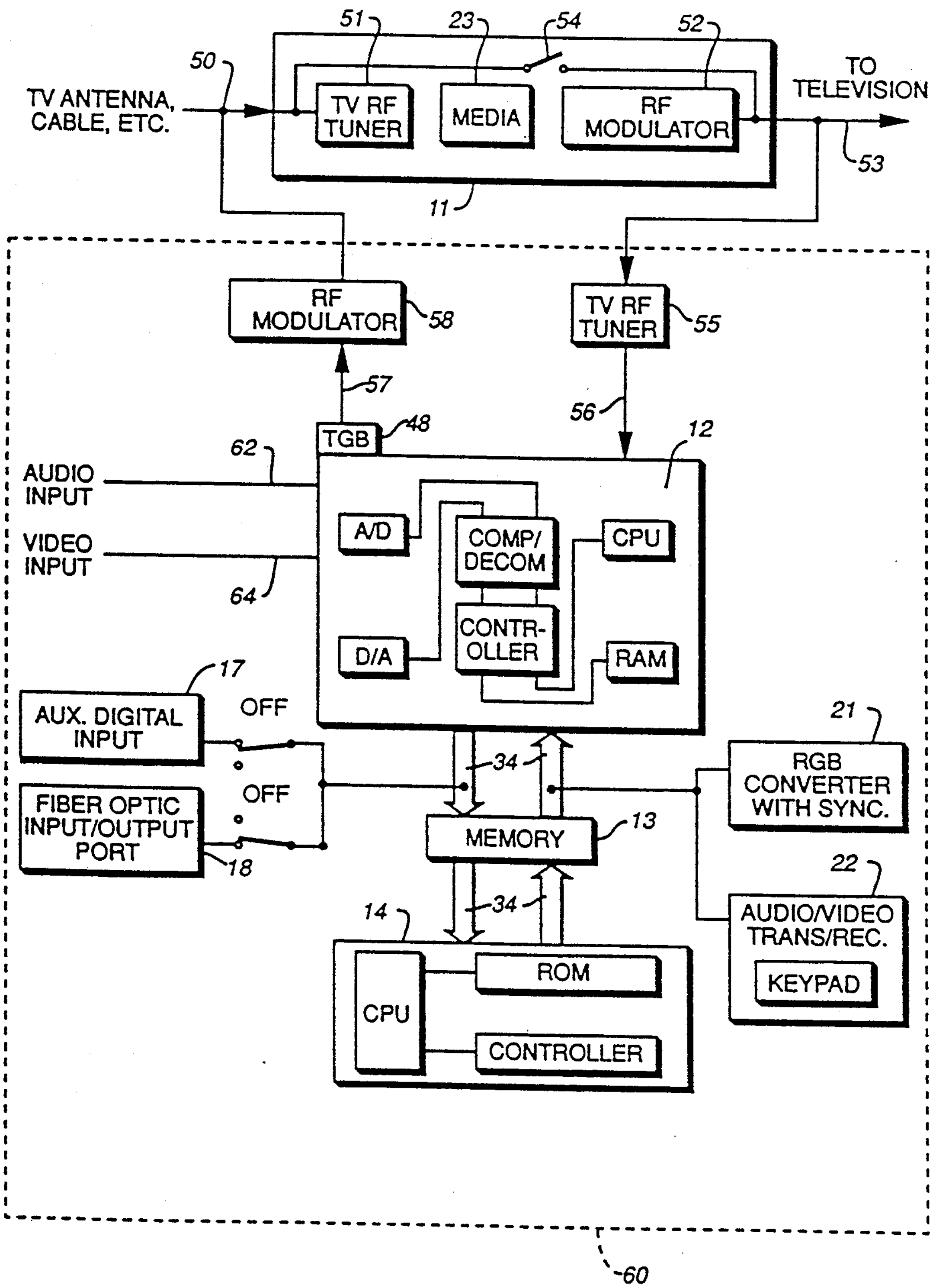
36
37
38
39
41
42
43
44
45
46
47
48
48'

FIG._2



51
23
54
52
50
TV ANTENNA, CABLE, ETC.
TV RF TUNER
MEDIA
RF MODULATOR
TO TELEVISION
53
11
RF MODULATOR
58
TV RF TUNER
55
57
56
TGB
48
12
62
AUDIO INPUT
VIDEO INPUT
64
A/D
COMP/ DECOM
CPU
D/A
CONTR-OLLER
RAM
17
AUX. DIGITAL INPUT
OFF
OFF
FIBER OPTIC INPUT/OUTPUT PORT
18
34
21
RGB CONVERTER WITH SYNC.
MEMORY
13
22
AUDIO/VIDEO TRANS/REC.
KEYPAD
14
34
CPU
ROM
CONTROLLER
60

FIG._3

AUDIO/VIDEO TRANSCEIVER APPARATUS INCLUDING COMPRESSION MEANS, RANDOM ACCESS STORAGE MEANS, AND MICROWAVE TRANSCEIVER MEANS

This Application is a continuation-in-part of my copending application Ser No. 07/289,776, filed Dec. 27, 1988, incorporated entirely, herein by reference and now U.S. Pat. No. 4,963,995.

BACKGROUND OF THE INVENTION

The video cassette recorder (VCR) has added significantly to the usefulness of the home television set. Important or exceptionally good programs may be recorded to be viewed again. Programs appearing at times that are inconvenient for viewing may be recorded for playback at a later time. Recorded movies or other materials, educational or entertaining, may be rented or borrowed for viewing at home. (As used in the remainder of this specification, the term "program" encompasses movies and other types of video and/or audio materials, whether broadcast from a TV station or another source.)

The typical VCR has its own tuner-receiver and a videorecorder It can receive and record a program from one channel while the television set is being employed to view a program on another channel. Programs are recorded on magnetic tape. The tape is then played back and viewed on the television set. Features commonly included in the VCR are capabilities for advancing the tape forward or backward at a high speed, stopping motion at any frame to hold the image, or simply playing back the recording at normal speed.

Desirable features that are not normally available in a VCR are capabilities for copying recorded programs from one tape or alternative storage medium to a similar or dissimilar storage medium, editing recorded programs and high speed recording. Another desirable, but currently unavailable, feature is the capability for high speed, high quality transmission and reception by optical fiber using the VCR.

DESCRIPTION OF THE PRIOR ART

U.S. Pat. No. 4,768,110, incorporated herein by reference, describes a VCR having two tape decks included therein. The purpose for the inclusion of two decks rather than the usual single tape deck is to permit the simultaneous viewing of a live RF-modulated TV signal or prerecorded material while recording another live RF-modulated TV signal and to also allow the copying of material from a first magnetic cassette tape onto a second magnetic cassette tape without the use of a second VCR. Viewing of the recorded material during the copying process is also possible in this arrangement. A major disadvantage is that the incorporation of the second tape deck is expensive and limited to magnetic tape, and furthermore, this prior art does not allow for the transmission or reception of recorded material over optical fibers or the high speed reception or transmission of audio/video material in a digital format. An additional disadvantage is the inability for random access editing of the audio/video signal. Furthermore, the additional mechanical structure adds significantly to the overall dimension of the equipment and increases the prospects of mechanical failures.

SUMMARY OF THE INVENTION

In accordance with the invention, an improved audio/video recorder is provided with added features and functions which significantly enhance its usefulness and functionality.

It is, therefore, an object of the present invention to provide an improved audio/video recorder for use in conjunction with an ordinary home television set.

Another object of the invention is to provide in such an improved audio/video recorder a capability for transferring a previously recorded program from one magnetic tape or other storage medium to another.

A further object of the invention is to provide such a capability for transferring a recorded audio/video program without resort to the use of two magnetic tape decks, this being a cumbersome, limited, and expensive approach already proposed in the prior art.

A still further object of the invention is to provide an effective and efficient means for intermediate storage of the audio/video program in digital memory as a means for achieving the transfer of the audio/video program from one tape or storage medium to another.

A still further object of the invention is to provide in such an improved audio/video recorder a capability for accepting various forms of analog or digital audio and video input signals and for converting the analog input signals to digital form when appropriate.

A still further object of the invention is to provide in such an improved audio/video recorder a capability for editing the video input signals without the necessity of using multiple cassettes or recording media.

A still further object of the invention is to provide an improved audio/video recorder for connection to various signal sources including a TV RF tuner, video camera, video and audio line input, and direct audio/video digital input from sources as diverse as a fiber optic input line, a microwave transceiver or a computer.

A still further object of the invention is to provide an improved audio/video recorder having a capability for mixing live audio/video programs with either analog or digital audio/video input signals from another source.

A still further object of the invention is to provide an improved audio/video recorder for simultaneously playing, viewing, recording and/or mixing digital and analog audio/video programs from different digital and analog audio/video sources or storage media.

A still further object of the invention is to provide an improved audio/video recorder which maximizes a given storage capacity, through the use of a data compression technique.

A still further object of the invention is to provide an audio/video recorder/transceiver utilizing a data compression technique for efficient storage of data, and efficient transmission and reception of a digitized audio/video program over a telephone line, a fiber optic cable, a microwave transceiver or other data transmission means.

A still further object of the invention is to provide in such an improved audio/video recorder a capability for delivering output signals in different forms or formats including a standard RF modulated output signal for viewing on a television set, a digital output signal for viewing on a high-resolution monitor, and audio output signals for a speaker system.

A still further object of this invention is to provide an improved audio/video recorder which provides for random access to any given segment of a self-stored

audio/video program so that the desired segment may be accessed and viewed without the time-consuming delays normally involved in fast-forward or fast-reverse searching procedures employed in present state-of-the-art VCR's.

A still further object of the invention is to provide an improved audio/video recorder which provides convenience in the editing of stored data by virtue of its random access memory capability.

A still further object of the invention is to provide an improved audio-video recorder which has the potential for enhanced audio and video quality by virtue of its capability for digital audio/video output and digital filtering techniques, and image or audio processing.

Further objects and advantages of the invention will become apparent as the following description proceeds, and the features of novelty which characterize the invention will be pointed out with particularity in the claims annexed to and forming a part of this specification.

BRIEF DESCRIPTION OF THE DRAWING

The present invention may be more readily described with reference to the accompanying drawing, in which:

FIG. **1** is a perspective view of the housing of the audio/video recorder editor/transceiver ("VCR-ET") disclosed and embodying the invention;

FIG. **1A** (is an enlarged view of the circled area of FIG. **1**;

FIG. **2** is a functional block diagram of the VCR-ET of FIG. **1**;

FIG. **3** is a functional block diagram of a VCR-ET in accordance with another embodiment of the invention; and

FIG. **4** is a functional block diagram of an audio recorder/transceiver constructed in accordance with the invention.

DESCRIPTION OF THE PREFERRED EMBODIMENT

Referring to the drawing by reference characters, FIGS. **1** and **2** illustrate an improved audio/video recorder editor/transceiver **10** (VCR-ET) comprising an audio/video recording unit (AVRU) **11**, a video control unit (VCU) **12**, memory **13**, digital control unit (DCU) **14**, video line or camera input line **15**, TV RF tuner **16**, auxiliary digital input port **17**, fiber optic input/output port **18**, RF modulator **19**, RGB converter with synchronizer **21**, and an audio/video transmitter/receiver **22** with keypad **45**, all in a common housing.

The audio/video recording unit AVRU **11** may be a video cassette recorder similar to a conventional VCR in which the storage media **23** is a magnetic tape. Alternatively AVRU **11** may operate with other types of storage media including, but not limited to, other magnetic tape formats. AVRU **11** has all the functions of the typical VCR including record, play, rewind, slow motion, fast-forward and single frame hold.

An alternate form of storage media for use in AVRU **11** is the CD-ROM, which is a disk using a derivative of glass or plastic in conjunction with an aluminum or other metallic coating Audio and video signals are stored in the form of irregularities in the aluminum coated surface and are read using a low power laser. In this case, the user would not be able to store or write on the CD-ROM, but would be able to play discs that have been recorded and distributed commercially. The storage of video and audio signals on the CD-ROM is in

digital form which is readily accommodated by the video recorder of this invention.

Instead of using a CD-ROM, VCR-ET **10** can use optical discs as media **23**. Such optical discs are similar to a CD-ROM and use a variable power laser to read from or write on the disc.

A first type of optical disc may comprise a WORM (Write Once Read Many) optical disc. This device has the unique capability of writing on the disc permanently. A laser is used to change the magnetic or optical properties of the media A lower-powered laser is then used to read the data from the disc. Data, in this case, is permanently recorded; it may neither be erased nor written over. A further description of this technology can be found in the Nov. 1988 issue of *The Electronic System Design* magazine (ESD) pages 55–56, incorporated herein by reference.

A second and preferred type of optical disc to be used in AVRU **11** is an erasable optical disc. This disc has full read/write/erase capabilities. With this disc, AVRU **11** has the same record/playback capabilities as a conventional VCR. As an example, erasable optical discs are used in Steven Jobs' "Next" machine as described in *Infoworld,* Volume 10, issue 42, pages 51 and 93, Oct. 17, 1988, incorporated herein by reference. In addition, the random access capabilities of the erasable disc (and of the CD-ROM and WORM) provide additional benefits as will be discussed in a later part of this specification.

A key element of VCR-ET **10**, which is responsible for its improved functionality, is the video control unit or VCU **12**. The VCU comprises an analog to digital converter (ADC) **24**, a digital to analog converter (DAC) **25**, a compressor/decompressor **26**, a controller **27**, a central processing unit (CPU) **28** and a random access memory (RAM) **29** VCU **12**, using these elements, accomplishes the digitization and compression of analog signals as well as the reverse process in which the compressed digital signals are decompressed and converted back to analog signals.

As a first step in the processing of the composite video signals within VCU **12**, the sync signals are decoded to isolate signals for each picture frame for processing.

The video signals defining each frame may then be converted to a red analog signal, a green analog signal, and a blue analog signal in a conventional manner. The red, green and blue analog signals are then converted to digital form by the analog to digital converter (ADC) **24**. The frame is divided into a set of closely positioned rows and columns of picture elements or "pixels." Each pixel has a color defined by a set of three digital values defining strength of the primary color components, red, green and blue (RGB) respectively. In one embodiment, each frame is divided into an array of 300 by 300 pixels, with the color and luminance of each pixel being defined by a seven bit word for the red component, a seven bit word for the blue component, and a seven bit word for the green component. These words are generated by ADC **24**. The RGB video signal may also be processed by means of hue-saturation-intensity (HSI) color processing, where appropriate, as described in "Chips for Real-Time Comparisons," *Electronic Engineering Times,* issue 525, Feb. 13, 1989, page 122.

If each frame includes 90,000 pixels (300×300), and each pixel is defined by 21 bits (7 bits per primary color), the digital representation of a single video frame utilizes a sizable block of digital information (i.e., 1.89 megabits/frame) which must be processed very rapidly.

(Approximately **30** frames/second are received from AVRU **11**). Fortunately the analog to digital conversion of these signals may be accomplished at the desired speed using commercially available analog to digital converter integrated circuits. The analog to digital converter **24** (ADC) is a high-speed, high-accuracy, A to D "flash" converter available as a single IC (integrated circuit). Several different types of such A/D converters are available from Burr-Brown, one of which is the ADC **600**. Part number TIC024, manufactured by Tektronix, Inc. is also appropriate. Other types of devices appropriate for this function are described in an article by K. Rogers entitled "8-bit A/D Flash Hits 500 Msamples", Electronic Engineering Times, Dec. 12, 1988, page 90, incorporated herein by reference.

Compression of the digital data defining a video frame and the reverse process (decompression) are accomplished by compressor/decompressor **26**. Various algorithms may be employed in the compression process which enable the representation of a series of numbers by a reduced number of digits As an example, compression algorithms like CCITT Group IV may be used.

In one optional embodiment, to further reduce the amount of memory required to store a program, the compression algorithm can simply record data corresponding to only those pixels which change color from one frame to the next. This results in considerable memory space savings, since not all pixels change color each frame. Basing calculation upon 10% of the pixels changing from one frame to the next, it is estimated that memory requirements using this technique are cut by about 90%. It is also estimated that on the average, the CCITT Group IV algorithm can cut memory requirements by another 95%. Thus, if no data compression technique is used, it would take approximately 51.03 gigabytes to store a 2 hour video program, but by using the above compression techniques, it is estimated that memory **13** will require only 250 megabytes.

Controller **27** handles timing and aids in the communication between the different elements of VCU **12**, and between VCU **12**, AVRU **11** and memory **13**.

In one embodiment, the audio portion of the program is periodically sampled and digitized by analog to digital conversion. In one embodiment, this is done at a sample rate of 88,000/second, one byte per sample, to yield compact disc quality sound. The sampling rate could be dropped to reduce memory requirements. Also, the audio data can be compressed with conventional algorithms.

The process of converting either from analog to digital or from digital to analog requires memory for intermediate storage. Random Access Memory (RAM) **29** serves in this capacity For this purpose either a DRAM (Dynamic RAM) or a SRAM (static RAM) may be employed. An example of a DRAM is the TI (Texas Instruments) TMX4C1024; an example of a SRAM is the INMOS IMS-1203. RAM **29** should have sufficient capacity to store at least two full uncompressed frames (e.g., about 472 KB).

The CPU (Central Processing Unit) **28** is a microprocessor which controls the digitization process of VCU **12**. CPU **28** works with controller **27** to control and communicate with the other elements of the VCU. There are numerous commercially available microprocessors that are appropriate for this application. The Intel 80286, Intel 80386, Motorola 68020, and Motorola 68030 are examples. A more complete description of the

microprocessors can be found in the Oct. 27, 1988 issue of *Electronic Design News* (EDN), pages 231 and 242, incorporated herein by reference, or in the applicable data sheets.

Controller **27**, CPU **28** and RAM **29** serve in the same manner during the reverse processes, i.e., decompression and digital to analog conversion. Decompression is first accomplished in compressor/decompressor **26**. The decompressed digital signal is then converted to an analog signal by digital to analog converter (DAC) **24** (assuming its destination requires an analog form). In the course of converting the decompressed signals from the VCU **12** for use by the AVRU **11** the signals are synchronized by the time base generator (TBG) or corrector **48**. TBG generator **48** inserts synchronization pulses into the signal provided by VCU **12** to identify individual raster scan lines and frames so that the resulting signal can be used by a conventional television set or VCR. TBG **48** can be bypassed by shunt switch **48'** for the purpose of transmitting either compressed or decompressed signals from VCU **12** directly to the AVRU **11** in an uncorrected time based mode.

DAC **25** provides the inverse of the function performed by A/D converter **24**. DAC **25** is a high-speed, high accuracy digital to analog converter. An example of such a converter is the Burr-Brown DAC60 digital to analog converter.

Different types of memory technologies are adaptable for use in memory **13**. As mentioned earlier, DRAM and SRAM semiconductor memories are commonly used for applications of this type and are readily available.

One type of random access memory is CMOS (Complimentary Metal Oxide Semiconductor). The CMOS memory has the advantage of a relatively low power requirement and is readily adaptable for use of battery backup for semipermanent data storage. Other types of memory include the above mentioned optical disc memories, bubble memories and magnetic disks. Another appropriate data storage media may be "Digital Paper" available from ICI Image data of Wilmington, Delaware.

Emerging memory technologies may also prove advantageous with capabilities for mass data storage in even smaller physical dimensions.

Digital Control Unit (DCU) **14** comprises a CPU (Central Processor Unit) **31**, a ROM (Read Only Memory) **32** and a controller **32**. DCU **14** is responsible for all of the digital editing processes. Through the use of DCU **14**, video segments may be edited and rearranged. Thus, one may use DCU **14** to rearrange the scenes in a program, alter the program sound track, etc.

In addition, a program may be edited, one frame at a time, by changing the contrast, brightness, sharpness, colors, etc. (Alteration of the contrast, brightness, sharpness and colors can be automated as well.) In one embodiment, images can be rotated, scaled (i.e., made larger or smaller), etc. In addition, pixel by pixel editing can be accomplished by DCU **14**, e.g., in a manner similar to a PC paint program. Similar editing features can be incorporated for the audio portion of each program. In one embodiment, a display such as a flat panel video display (not shown) is built into the VCR-ET. A user interface control panel of DCU **14** allows a user to select a desired frame number from a menu on the display. The VCR-ET then displays a strip of frames (including several frames before and after the selected frame). The user can delete frames in a strip, select a

point where other frames are to be inserted into the program, or edit different frames (i.e., alter contrast, brightness, sharpness, colors, etc.). In one embodiment, a user input device such as a light pen or mouse can be used to select individual frames in a strip for editing.

Instead of incorporating a flat display into VCR-ET **10**, in another embodiment, a television coupled to output lead **42** of RF modulator **19** can be used during editing.

CPU **31** is a microprocessor of the type described in connection with the CPU **28** of VCU **12**. Controller **33** is an integrated circuit which handles the timing and interfacing between DCU **14** and memory **13**. ROM **32** holds the necessary step-by-step editing programs which are installed at the factory A currently available example of a suitable ROM for this application is the Texas Instruments part TMS47256 CPU **31** and controller **33** together control the editing process as they execute the programs stored in ROM **32**.

The VCU **12**, memory **13** and DCU **14** communicate with each other via a high speed data bus **34**. The high speed data bus is required in order to meet bandwidth requirements. Examples of suitable data bus devices are Motorola's VME bus, Intel's Multibus and the Optobuss (U.S. Pat. No. 4,732,446).

A video line or camera input line **15** is provided to enable VCR-ET **10** to receive an input signal from a source such as a television camera, a conventional VCR, a television tuner, or another VCR, etc. The signals received at input line **15** are typically carried by a coaxial cable and are in the form of a standard television composite signal. As used throughout this specification, the words "standard television composite signal" or its acronym STCS shall be read to include any one of the following: NTSC, PAL, SECAM, HDTV, or any American or European broadcast signal standards. (NTSC, PAL and SECAM are discussed in "Reference Data for Radio Engineers", published by Howard W. Sams & Co. in 1983, incorporated herein by reference.) An NTSC composite signal is defined as the analog signal that carries the chrominance (color), luminance (brightness), synchronization (timing) and audio signals that make up the video signals received and displayed by television and video cassette recorders. These four components are combined into one signal by modulating the components in different ways. (Amplitude modulation and phase modulation are examples.) The standard video line signal is such a composite signal and may be received at input line **15** from one of the above-mentioned sources.

TV RF tuner input port **16** also supplies a composite signal as described in regard to video input line **15**. The difference is that this signal is received from an antenna or cable TV coaxial cable. To receive such a signal, tuner **16** is capable of being set or tuned to receive the desired carrier frequency or television channel.

Selector switch **35** is provided to select either video input line **15** or TV RF tuner **16** as an input signal source to AVRU **11**.

Auxiliary digital input port **17** is employed to receive any acceptable digital signal such as computer-generated video signal or as may be supplied by another VCR-ET. This signal, for example, may be an RGB video signal such as that delivered to computer monitors, or it may be a digitized audio signal. (As mentioned above, an RGB signal is a signal which communicates the strength of the red, green and blue color components for the pixels that make up each video frame.)

Switch **36** selects whether the digital video/audio input signal is chosen from auxiliary digital input port **17**. Switch **36** supplies the selected signal to high speed data bus **34** which carries the signals in digital form.

Fiber optic port **18** incorporates a fiber optic transceiver Port **18** has a capability for transforming fiber optic (light) signals to electrical signals or for transforming electrical signals to fiber optic signals Port **18** thus provides a capability for two-way communication between high speed data bus **34** and a fiber optic signal line. The incorporation of fiber optic port 18 in the VCR-ET provides a capability for receiving audio/video signals from or delivering audio/video signals to the fiber optic line such as a fiber optic telephone line. The fiber optic line carries digital signals in the form of light waves over great distances with a high degree of accuracy and reliability and at a high speed (e.g., about **200** megabytes/second). The VCR-ET can receive/transmit a video program at an accelerated rate via fiber optic port **18** from/to a variety of sources. For example a video program may be communicated at an accelerated rate from the first VCR-ET to a second VCR-ET in less time than it would take to view the program. Thus, it is not necessary to access the optical fiber for long periods of time in order to transmit a long video program.

It is also envisioned that in the future, a video library may be established which downloads video programs at an accelerated rate via optical fibers to a subscriber's VCR-ET. After downloading, the program may be viewed, stored in memory, edited and/or a hard copy of the program may be made on magnetic tape, optical disk, etc.

Switch **37** is provided to select connection to the fiber optic input/output port **18**. An OFF or open position is provided The selected signal is delivered to or supplied from high speed data bus **34**.

Analog output signals from AVRU **11** are delivered to the common terminal **38** of a selector switch **39**. When set to position A, switch **39** delivers the output signal of AVRU **11** directly to a video output line **41** as a standard STCS composite signal; when set to position B switch **39** delivers the output of VRU **11** to the input of RF modulator **19**. Modulator **19** converts the video signal to an RF-modulated composite signal for delivery to such devices as televisions and conventional VCR's. These types of devices play back the video program on a particular frequency channel (such as channel **4**) on the television. Delivery to the television or VCR is via RF output line **42**.

Digital output signals from VCR-ET **10** may be dispatched from high speed data bus **34** via line 43 to input leads of RGB converter **21** and audio/video transmitter/receiver **22**.

RGB converter **21** converts the STCS signal into an RGB signal as required by computer monitors and similar display devices. The converted signal is received by a display device connected to RGB converter output line **44**.

VCR-ET **10** includes audio/video transmitter/receiver **22** which is typically a built-in modem. Advantageously, the modem may be used to communicate an audio/video program over conventional phone lines in a manner similar to that described above with respect to optical fibers. The term modem is derived directly from its functionality as a modulator-demodulator which allows transfer of the audio/video signal in a digital format over the standard telephone line. Modems are

commonly available for computers and are currently available in the form of a single integrated circuit. As an example, Sierra Semiconductor offers a 2400 baud single chip modem under its part number SC111006. Representative manufacturers of these single modem IC's can be found in the Apr. 14, 1988 issue of

Engineering Design News (EDN), pages 124–125. Some of these single IC modems have the added capability of generating the tones for dialing a phone number. The destination phone number may be entered by means of an optional keyboard/keypad **45** incorporated in the video recorder **10** of the invention. Output port **46** of transmitter/receiver **22** connects directly to the telephone line.

Also associated with Modem **22** is an auxiliary keyboard **45'** (FIG. 1A) of buttons for commanding the modem to perform tasks such as starting a transmission over phone lines (**45***a*), terminating a transmission (**45***b*), automatic telephone answering to receive transmissions (**45***c*), using an optional speaker (not shown) to monitor phone lines (**45***d*), using an optional microphone (not shown) to speak over the phone lines (**45***e*) and for controlling the baud rate (**45***f*).

The application and utilization of the VCR-ET may include a number of forms or operating modes. In its first and simplest operating mode, AVRU **11** may be operated in the manner of a conventional VCR with signals from an antenna being received by tuner **16** and recorded directly on media **23** in analog form. At the same time the received program may be viewed on the television screen with the television connected at video output terminal **42**. An optional signal source for this type of operation is the video line or camera input line **15** selectable by switch **35**.

In a second operating mode a program stored in media **23** of AVRU **11** may be played back and viewed on the connected television set.

When it is desired to copy a program from one recording media to another, the recording media holding the desired program is installed in the AVRU. The recording media is then played back with optional viewing on a connected television set or other TV monitor or listening through speakers (as appropriate). As the recording media is played back, the analog signals from the recording media (video and/or audio) are dispatched to VCU **12** via connection **47**. The analog signals are converted to digital signals by ADC **24**, compressed by compressor/decompressor **26** and the compressed digital signals are stored in memory **13**. The foregoing operations are accomplished under the control of controller **27** and CPU **28**. RAM **29** is used for interim data storage during this process. Once the complete video/audio program has been stored in memory **13**, the recording media from which the stored program has just been read is replaced by blank recording media upon which the stored program is to be copied. CPU **28** in cooperation with controller **27** and RAM **29** then executes the decompression and digital to analog conversion of the program stored in memory **13**, decompression taking place in compressor/decompressor **26**, and digital to analog conversion being accomplished by DAC **25**. The resulting analog program is stored on the blank recording media which constitutes media **23** of AVRU **11**.

In an alternate mode of operation, the decompression circuitry of VCU **12** can be bypassed. Thus, a user has the option of downloading the stored program from memory **13** onto recording media **23** in compressed digital format. The user can then reload the program from media **23** into memory **13** at a future time for viewing, editing or recording back onto recording media **23** in analog form. This capability allows the user to quickly clear memory **13** for other interim uses and also provides the user with a hard copy of the program in digital format. The hard copy in compressed digital format has a number of uses, e.g. it could be archived for later viewing, transmitted by an appropriate independent transmitter, etc.

During the foregoing procedures, DCU **14** may be utilized for editing operations. As the program is being read from the first or original recording media, it is simultaneously viewed on the TV screen, or listened to by means of an audio monitor, converted to digital signals, compressed and stored in memory **13**. Once the digital audio/video program is stored in memory **13**, editing is accomplished by the user through control of DCU **14**, by means of a control panel (not shown) coupled to DCU **14**. If desired, additional audio/video signals may be simultaneously entered into memory **13** and added to those received from VCU **12**. The additional signals may be introduced from auxiliary digital input port **17** or from fiber optic input/output port **18** and may comprise video captions for super imposed position upon the stored video images, or they may be audio commentaries to be added to silent video presentations. In addition, as mentioned above, the order in which various segments appear in the video programs may be altered. Certain undesired segments, such as TV commercials, may be removed. This editing operation is accomplished under the control of DCU **14**.

In still another operating mode, a program stored in media **23** of AVRU **11** or being received by AVRU **11** from input line **15** (as from a video camera) may be digitized and compressed by VCU **12** and routed via bus **34**, to memory **13**. The data from memory **13** is then routed to line **43**, transmitter/ receiver **22** and to a telephone line. At the other end of the telephone line the signals received are processed by another VCR-ET.

Once received in the second VCR-ET's memory **13**, the digitized program can then either be viewed directly from memory or transferred to storage medium **23**, either in its entirety or in random segments, based on user preference.

In the case of video camera input at input **15** the transmitted signals may comprise a live transmission. Alternatively the transmitted program may be derived from a program stored in media **23** of AVRU **11**. In this case the stored analog program is again decoded, digitized, compressed and transmitted via bus **34** to memory **13**. The data in memory **13** is then communicated via line **43** and transmitter/receiver **22** to telephone lines.

It follows, of course, that digitized video and audio signals from the remote VCR-ET at the other end of the telephone line may be received at line **46**, entered into memory **13** via transmitter/receiver **22**, converted to analog signals by VCU **12**, and recorded on media **23** and then viewed, if desired, on a television set connected at output **42**. A hard copy of the program may also be made for later viewing.

As mentioned earlier, when any of the foregoing operations entail the processing of unmodulated video signals, such signals must first be processed by RF modulator **19** before they can be accepted by devices such as a conventional VCR or television set; when the monitoring means is a computer monitor or a similar display device the signals are processed by RGB converter **21**.

All of the foregoing operations are performed with enhanded quality and efficiency by virtue of the digital, rather than analog, storage and transmission modes and the compressed data storage mechanism, with additional advantages of improved cost and reliability afforded in the case of tape to tape (or other media to media) program transfers by virtue of the requirement for only a single tape deck or other storage device.

FIG. **3** illustrates an alternative embodiment invention in which AVRU **11** is not integral with VCU **12**, memory **13** or editor **14**. In this embodiment, AVRU **11** is a conventional, commercially available VCR which receives a modulated video input signal on an input cable **50**. In this embodiment AVRU **11** includes a RF tuner **51** for demodulating the input signal so it can be stored in media **23**. AVRU **11** also includes a RF modulator **52** for modulating the signal received from media **23** and providing the RF modulated output signal on an output cable **53**, which can be coupled to a television set. (TV RF tuner **51** and RF modulator **52** are provided in typical commercially available VCR's.) A switch **54** is provided to couple input cable **50** to output cable **53** when media **23** is not serving as a video signal source. The VCR-ET of this embodiment includes a TV RF tuner **55** which receives and demodulates the signal on cable **53**, and provides the resultant analog audio/video signal on a lead **56**, which is digitized and compressed as described above. In this alternative embodiment, the digitized compressed signal may be processed as described above, e.g. stored in memory **13** (via high speed bus **34**), edited, transmitted by the fiber optic port **18** to another VCR-ET, etc. When it is desired to view a program stored in memory **13**, data from memory **13** is decompressed and converted to an analog signal by VCU **12**, and the resulting signal is provided on an output lead **57** to a RF modulator **58**, which modulates the video signal so that it can be received and stored by AVRU **11** or viewed on a television coupled to cable **53**. (As mentioned above, in the FIG. **3** embodiment, AVRU **11** is a conventional VCR.)

One advantage of the embodiment of FIG. **3** is that many people already own VCR's. Rather than buying apparatus which duplicates much of the hardware already present in their VCR, the embodiment of FIG. **3** would provide to owners of conventional VCR's capabilities which are otherwise currently unavailable in an economical manner.

In one embodiment, analog auxiliary audio and video input terminals **62, 64** are provided so that analog signals may be provided by alternate sources to VCU **12**.

The embodiments described above include means for transmitting/receiving video programs over fiber optic cables However, in an alternative embodiment, either in place of fiber optic port **18** or in addition to fiber optic port **18**, means are provided for transmitting and/or receiving a video program via microwave. In conventional microwave technology, satellite systems and microwave transmitters transmit data using a low power/high frequency signal. In an embodiment of the invention designed to receive microwaves, the VCR-ET includes an amplifier for amplifying the microwave signal and a demodulator for obtaining the video program signal from the microwave signal. Receiving, amplifying and demodulating the microwave signal can be accomplished with conventional microwave transceiver equipment. The video program signal is typically in digital form, and may be stored, viewed or edited as in the above-described embodiments. Program data from memory **13** can also be transmitted by the microwave transceiver, thereby providing the capability for microwave transmission of stored video programs in compressed digital format. Thus, the invention can be used to receive and transmit programs via microwaves at an accelerated rate similar to and at least as fast as, the transmission and reception of programs over optical fibers. This feature allows transmission and reception of programs in a few minutes or seconds using currently available technology. Both point-to-point microwave transceivers and satellite transceivers may be used.

The embodiments described include means for receiving, storing and transmitting both audio and video signals. However, the invention encompasses apparatus which can store and transmit video signals only and apparatus which can store and transmit audio signals only. An embodiment designed to store and compress audio signals is illustrated in FIG. **4**. Referring to FIG. **4**, an audio signal source **70** (a tape recorder, microphone, record player, etc.) is coupled to a digitizer and compressor circuit **72**, which converts the analog signal to a digital signal and compresses the digital signal in a manner similar to VCU **12** described above. The digital compressed signal can then be stored in a memory **74**. Of importance, data from memory **74** can be transmitted by a fiber optic transceiver **76**, or by a microwave transceiver **78** at an accelerated rate. This is important not only in a home entertainment application, but in other applications as well. For example, a user can dictate an audio presentation and send it to a remote location (e.g. an office) at an accelerated rate without having to monopolize the transmission medium (e.g. the fiber optic cable) for an extended length of time.

The business uses of the embodiment illustrated in FIG. **4** makes home offices feasible for many workers now confined to more traditional offices and also opens new possibilities to business people who are traveling.

In the embodiment of FIG. **4**, data can also be loaded from memory **74**, via a modem **79** over a conventional phone line **80**. Data can also be received from phone line **80**, fiber optic transceiver **76** or microwave transceiver **78**, loaded into memory **74**, and converted to an analog signal by circuit **72**, to be listened to via an audio monitor **82**, or to be stored on an audio tape cassette **84** or other storage media.

An editor **86** is optionally provided so that the data in memory **74** may be edited, e.g., by rearranging the order of portions of the audio program, increasing or decreasing the volume of portions (or different frequency components) of the audio program, or enhancing the audio program through filtering techniques (e.g. to remove static and noise).

An improved audio/video recorder with significantly expanded functional capabilities is thus provided in accordance with the stated objects of the invention and although but a single embodiment of the invention has been illustrated and described, it will be apparent to those skilled in the art that various changes and modifications may be made therein without departing from the spirit of the invention or from the scope of the appended claim. For example, the VCR-ET can be constructed so as to be portable. Thus, it could be carried to a location where it is desired to record a program, and used to edit the program after it is recorded with a video camera. Other modifications will be apparent to those skilled in the art in light of the present specification.

What is claimed is:

1. An audio/video transceiver apparatus comprising:

input means for receiving audio/video source information, said audio/video source information comprising a multiplicity of video frames in the form of one or more full motion video programs;

compression means, coupled to said input means, for compressing said audio/video source information into a time compressed representation thereof having an associated time period that is shorter than a time period associated with a real time representation of said audio/video source information;

random access storage means, coupled to said compression means, for storing the time compressed representation of said audio/video source information; and

output means, coupled to said random access storage means, for receiving the time compressed audio/video source information stored in said random access storage means for transmission away from said audio/video transceiver apparatus;

said input and output means comprising microwave transceiver means, coupled to a microwave link, for receiving said audio/video source information over said microwave link and for transmitting said time compressed audio/video source information stored in said random access storage means over said microwave link.

2. An audio/video transceiver apparatus comprising:

input means for receiving audio/video source information, said audio/video source information comprising a multiplicity of video frames in the form of one or more full motion video programs;

compression means, coupled to said input means, for compressing said audio/video source information into a time compressed representation thereof having an associated time period that is shorter than a time period associated with a real time representation f said audio/video source information;

random access storage means, coupled to said compression means, for storing the time compressed representation of said audio/video source information, said random access storage means comprising a bubble memory; and

output means, coupled to said random access storage means, for receiving the time compressed audio/video source information stored in said random access storage means for transmission away from said audio/video transceiver apparatus.

3. An audio/video transceiver apparatus comprising:

input means for receiving audio/video source information, said audio/video source information comprising a multiplicity of video frames in the form of one or more full motion video programs;

compression means, coupled to said input means, for compressing said audio/video source information into a time compressed representation thereof having an associated time period that is shorter than a time period associated with a real time representation of said audio/video source information;

random access storage means, coupled to said compression means, for storing the time compressed representation of said audio/video source information, said random access storage means comprising digital paper; and

output means, coupled to said random access storage means, for receiving the time compressed audio/video source information stored in said random access storage means for transmission away from said audio/video transceiver apparatus.

4. An audio/video transceiver apparatus comprising:

input means for receiving audio/video source information, said audio/video source information comprising a multiplicity of video frames in the form of one or more full motion video programs;

compression means, coupled to said input means, for compressing said audio/video source information into a time compressed representation thereof having an associated time period that is shorter than a time period associated wth a real time representation of said audio/video source information;

random access storage means, coupled to said compression means, for storing the time compressed representation of said audio/video source information, said random access storage means comprising one or magnetic disks; and

output means, coupled to said random access storage means, for receiving the time compressed audio/video source information stored in said random access storage means for transmission away from said audio/video transceiver apparatus;

5. An audio/video transceiver apparatus comprising:

input means for receiving audio/video source information as a time compressed digital representation thereof, said audio/video source information comprising a multiplicity of video frames in the form of one or moore full motion video programs, said time compressed digital representation of said audio/video source information being received over an associated burst time period that is shorter than a real period associated with said audio/video source information;

random access storage means, coupled to said input means, for storing the time compressed digital representation of said audio/video source information received by said input means; and

output means, coupled to said random access storage means, for receiving the time compressed digital representation of said audio/video source information stored in said random access storage means for transmission away from said audio/video transceiver apparatus;

said input and output means comprising microwave transceiver means coupled, via a microwave link, to a video library, said video library storing a multiplicity of full motion video programs in said time compressed digital representation for selective retrieval, inn said associated burst time period, over said microwave link, said microwave transceiver means being further operative for transmitting in said burst time period, said time compressed digital representation of said audio/video source information stored in said random access storage means over said microwave link.

* * * * *

EXHIBIT E

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United States Patent [19]

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Polish

[54] **SYSTEM AND METHOD FOR DISTRIBUTING AND MANAGING DIGITAL VIDEO INFORMATION IN A VIDEO DISTRIBUTION NETWORK**

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[52] **U.S. Cl.** **345/327**; 348/7

[58] **Field of Search** 348/7, 12, 13, 348/10, 11; 455/4.2, 5.1, 6.2, 6.3; 345/327; 395/200.49; H04N 7/16, 7/173

[56] **References Cited**

U.S. PATENT DOCUMENTS

5,371,532 12/1994 Gelman et al. 348/7
5,710,970 1/1998 Walters et al. 348/7
5,721,815 2/1998 Ottesen et al. 348/7
5,729,280 3/1998 Inoue et al. 348/7
5,737,009 4/1998 Payton 348/7
5,742,347 4/1998 Kandlur et al. 348/7
5,815,194 9/1998 Ueda 348/7

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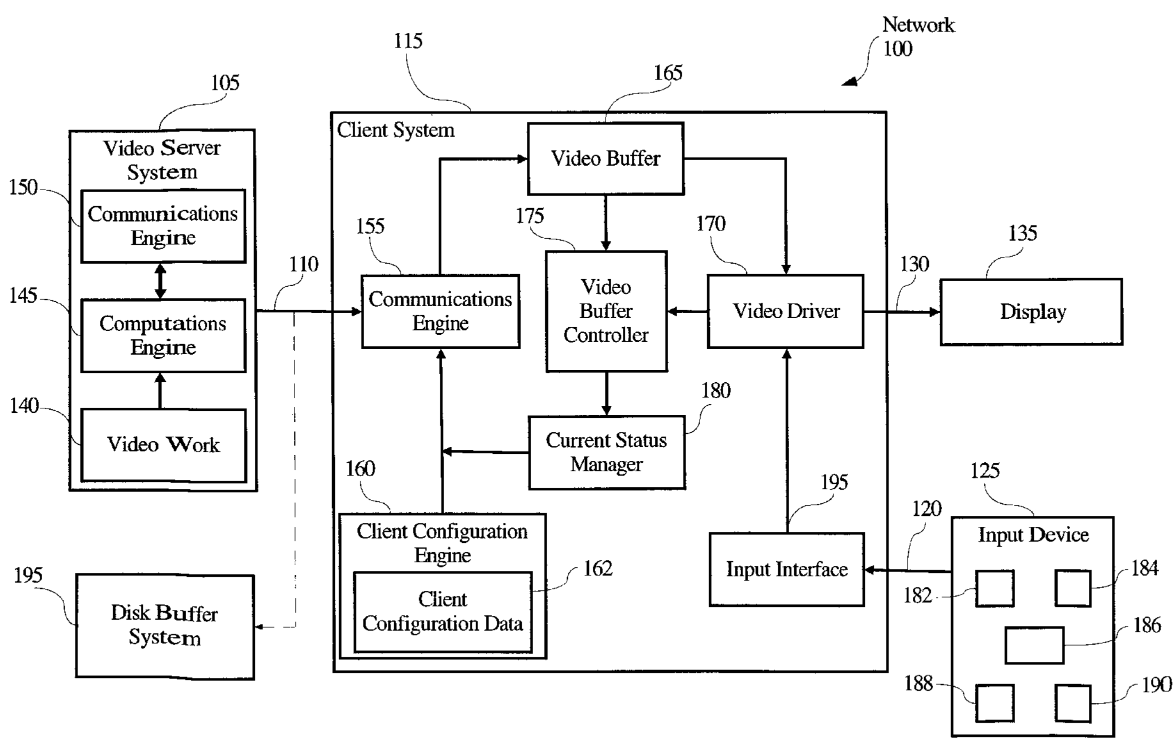
Attorney, Agent, or Firm—Carr & Ferrell LLP

[57] **ABSTRACT**

The video distribution network system includes client configuration data, a client video buffer for storing video information, a client video driver coupled to the client video buffer for presenting a portion of the video information on a display device, a current status manager for determining current client status information indicative of the portion of video information presented, a computations engine coupled to the client video buffer and to the current status manager for forwarding a burst of video information to the client video buffer based on the client configuration data and on the client status information, and a video buffer controller coupled to the client video buffer for controlling storage of the burst in the client video buffer.

47 Claims, 8 Drawing Sheets



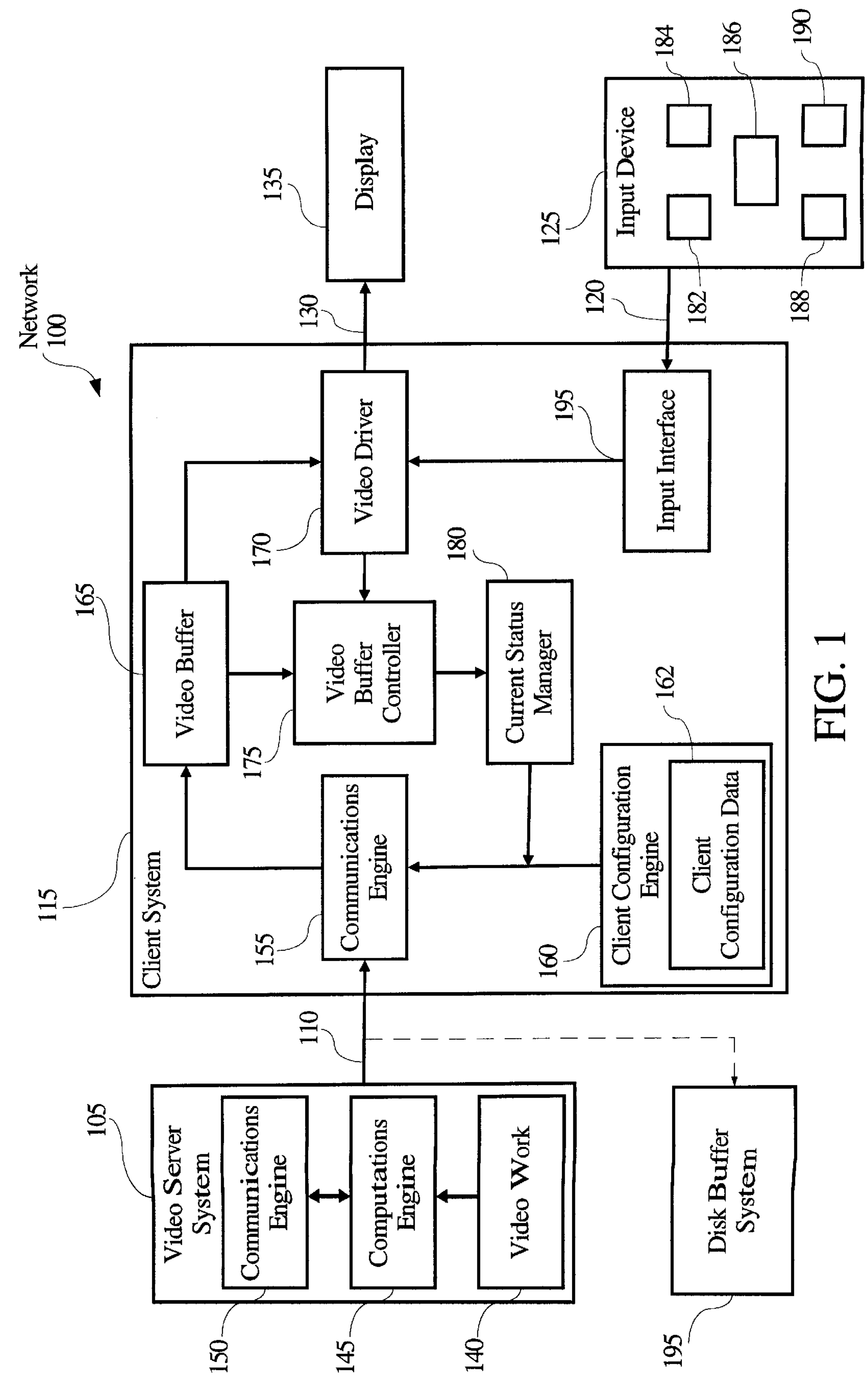

Network
100
Video Server System
105
Communications Engine
150
Computations Engine
145
Video Work
140
110
Disk Buffer System
195
Client System
115
Communications Engine
155
Video Buffer
165
Video Buffer Controller
175
Video Driver
170
Current Status Manager
180
Client Configuration Engine
160
Client Configuration Data
162
Input Interface
195
130
Display
135
120
Input Device
125
182
184
186
188
190

FIG. 1

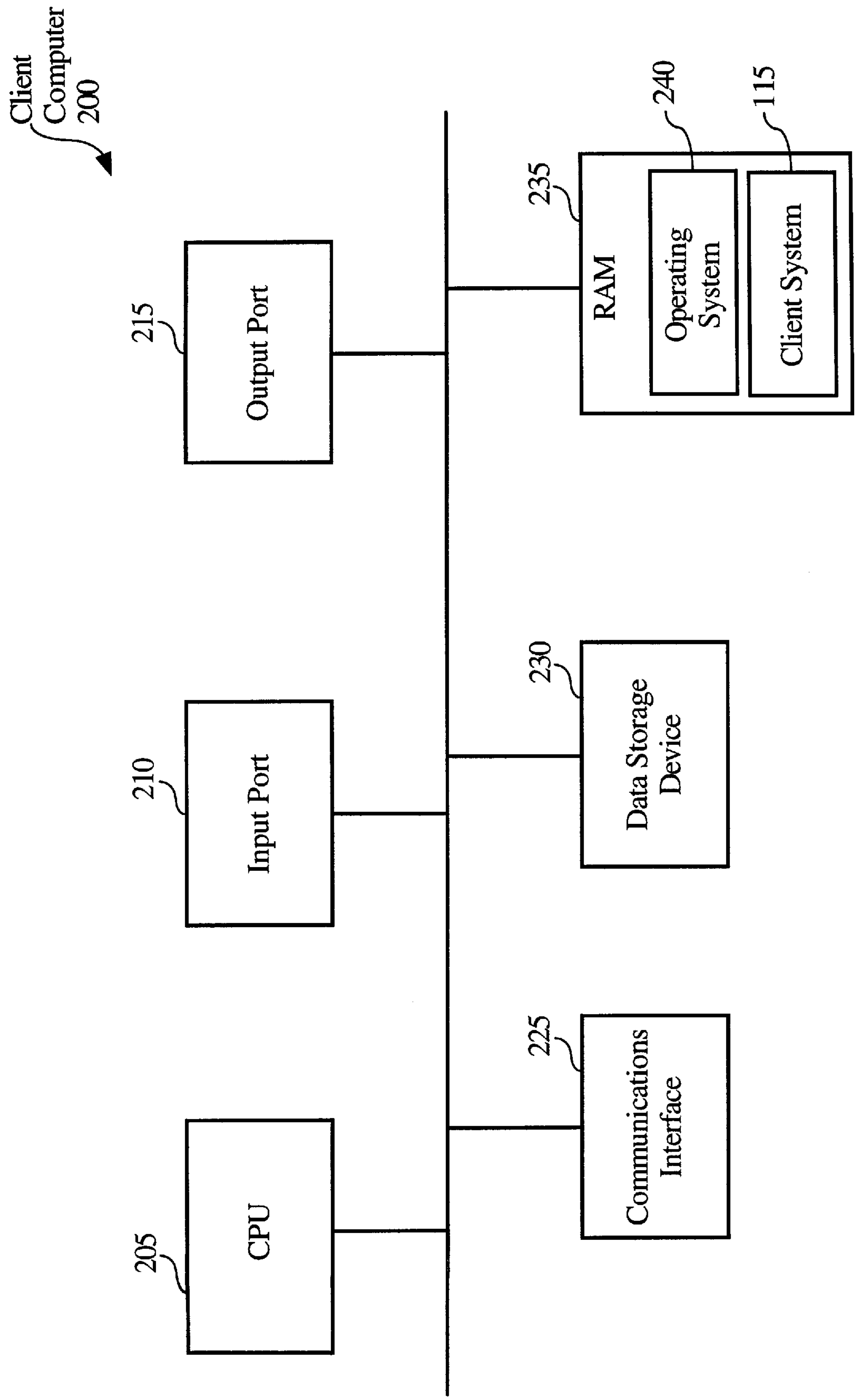

Client Computer 200
215
Output Port
210
Input Port
205
CPU
235
RAM
240
Operating System
115
Client System
230
Data Storage Device
225
Communications Interface

FIG. 2

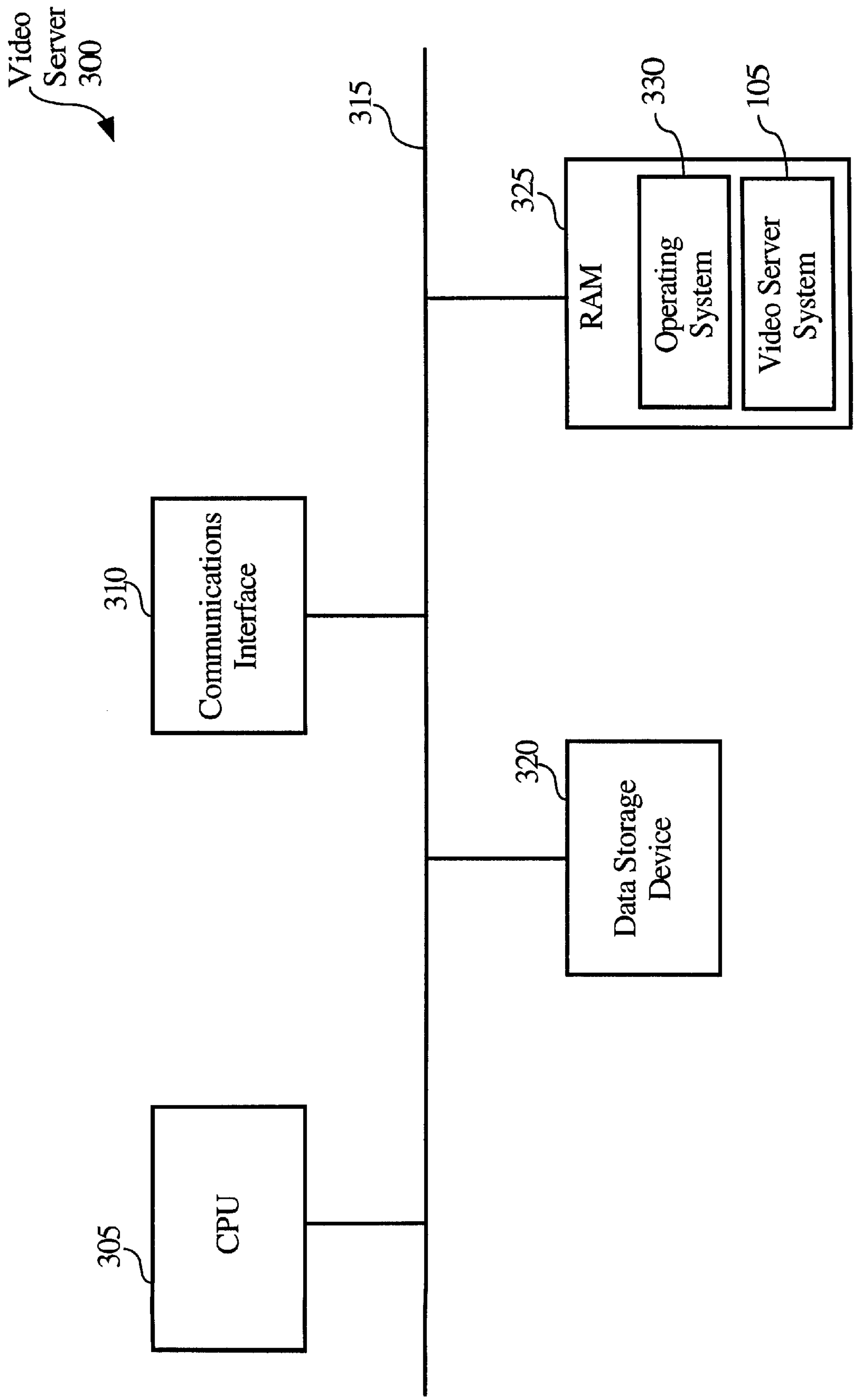

Video Server 300
315
325
RAM
330
Operating System
105
Video Server System
310
Communications Interface
320
Data Storage Device
305
CPU

FIG. 3

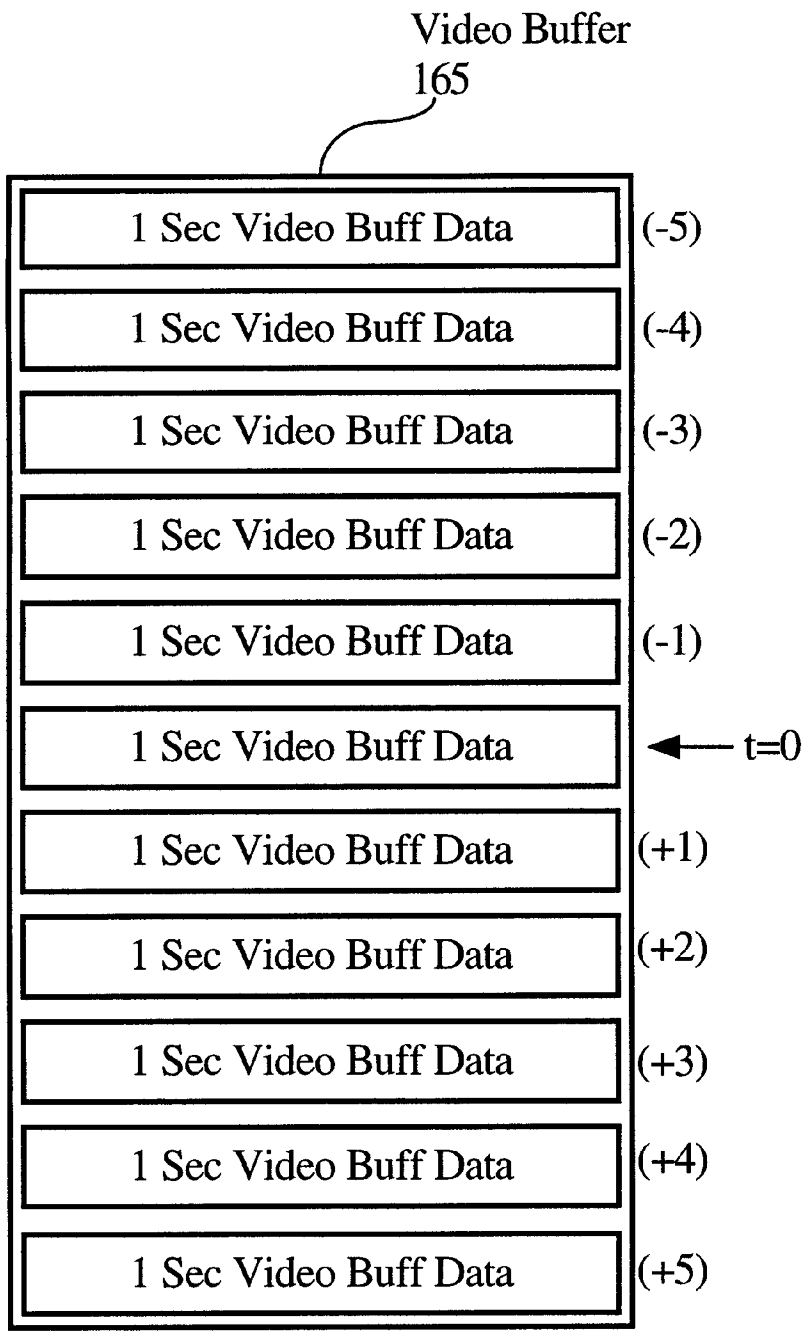

Video Buffer
165
1 Sec Video Buff Data
(-5)
1 Sec Video Buff Data
(-4)
1 Sec Video Buff Data
(-3)
1 Sec Video Buff Data
(-2)
1 Sec Video Buff Data
(-1)
1 Sec Video Buff Data
t=0
1 Sec Video Buff Data
(+1)
1 Sec Video Buff Data
(+2)
1 Sec Video Buff Data
(+3)
1 Sec Video Buff Data
(+4)
1 Sec Video Buff Data
(+5)

FIG. 4

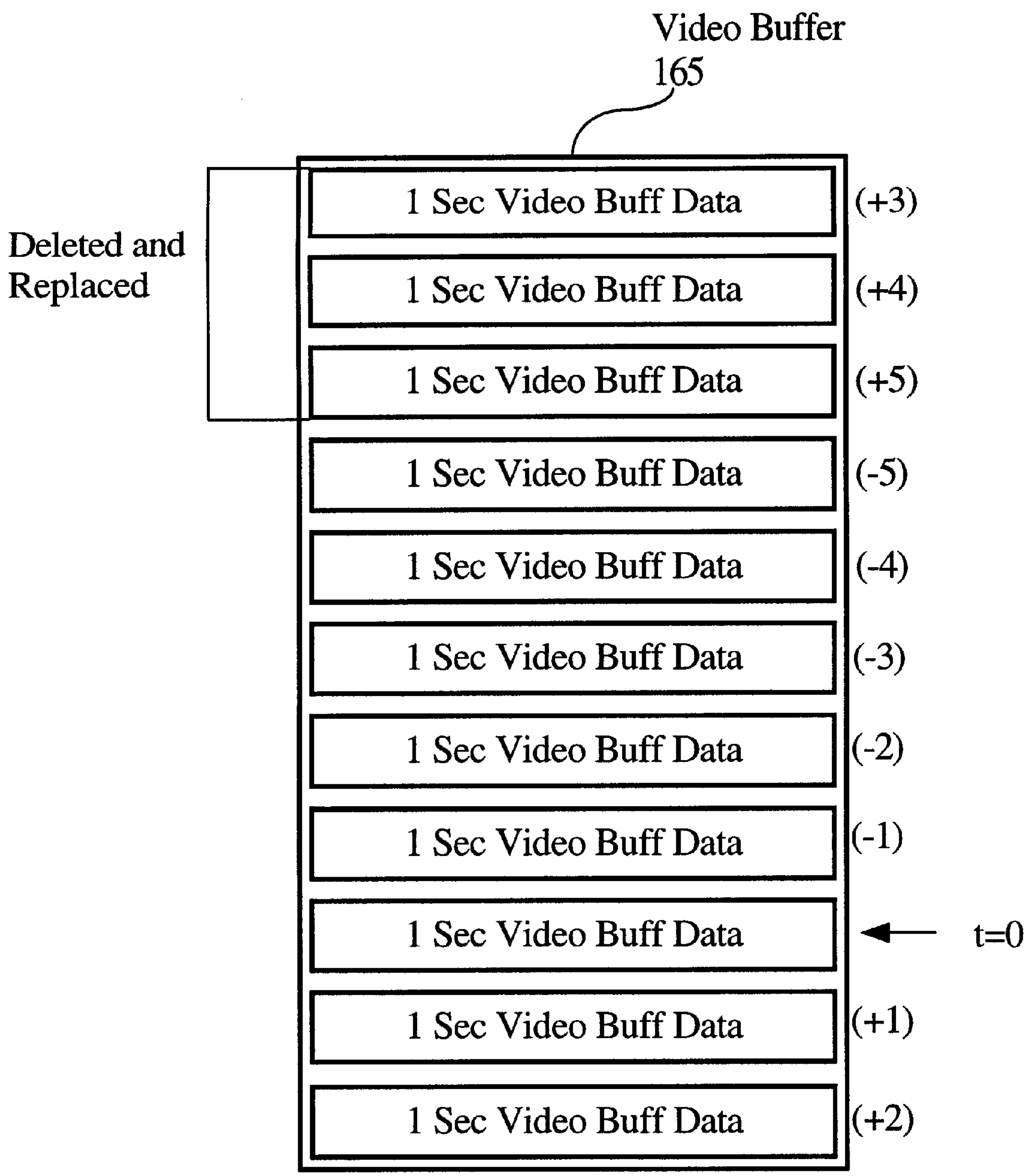


FIG. 5
(3 secs of play)

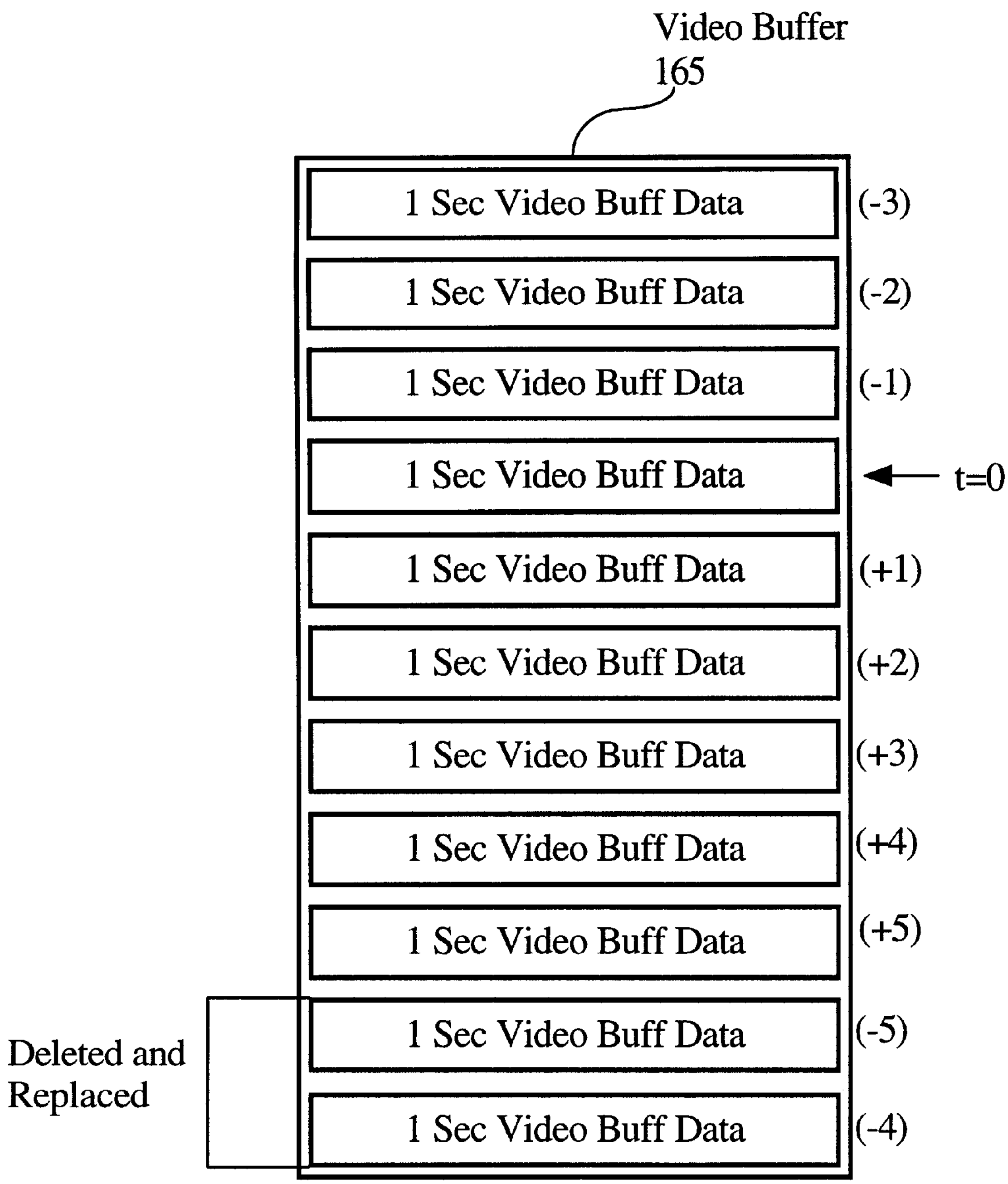


FIG. 6
(2 secs of rewind)

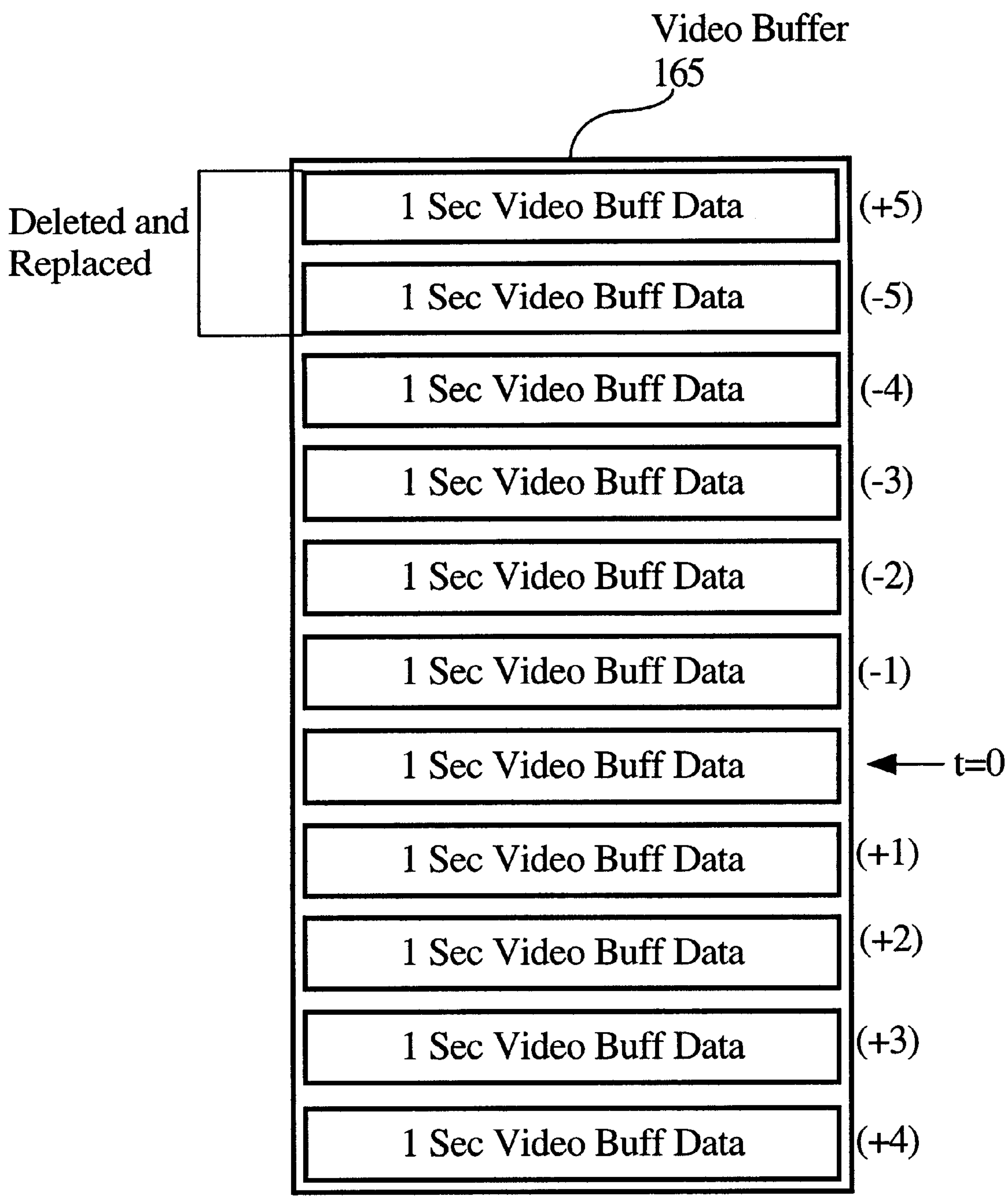


FIG. 7
(3 secs of play and 2 secs of rewind)

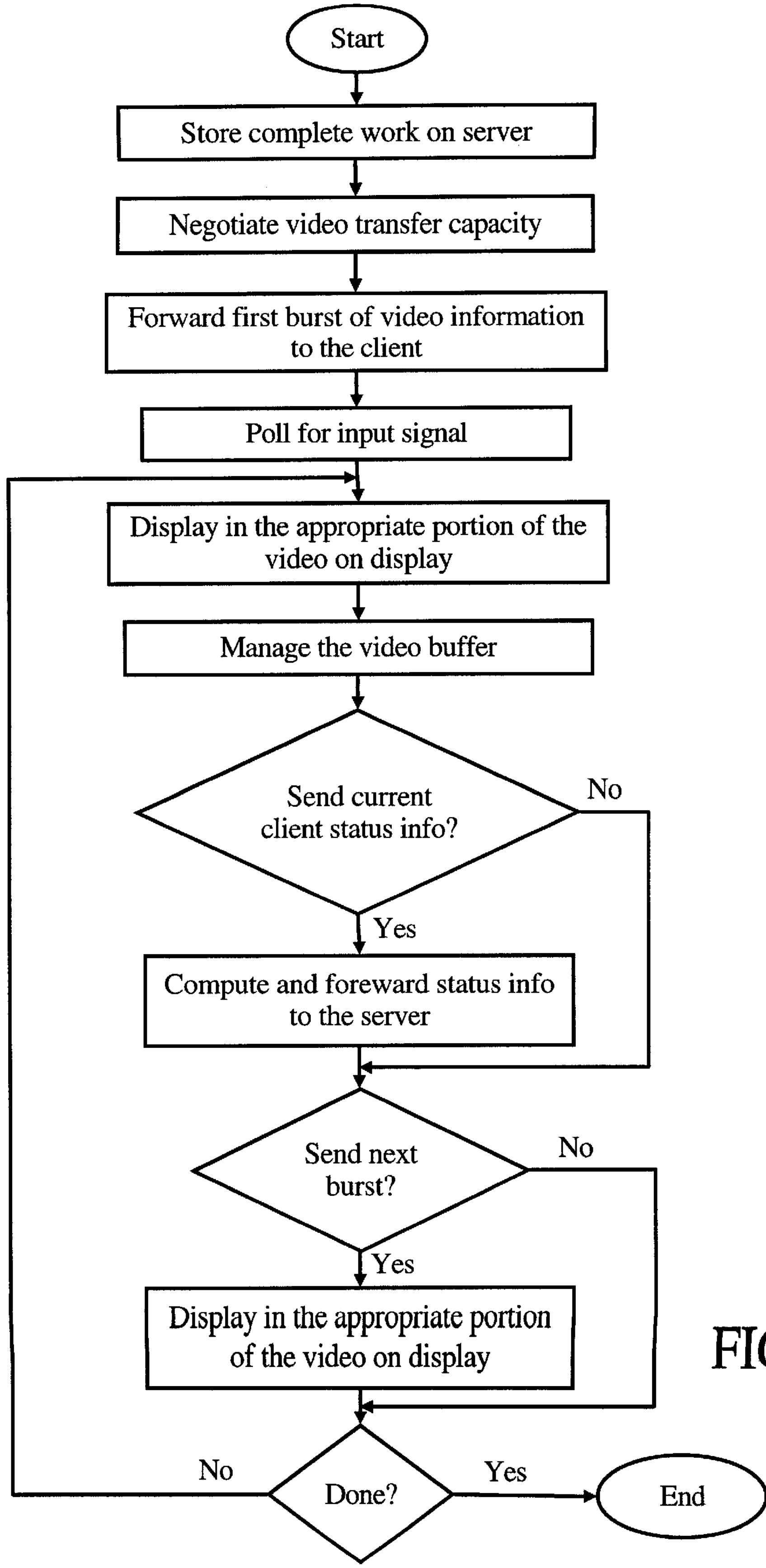

Start
Store complete work on server
Negotiate video transfer capacity
Forward first burst of video information to the client
Poll for input signal
Display in the appropriate portion of the video on display
Manage the video buffer
Send current client status info?
No
Yes
Compute and foreward status info to the server
Send next burst?
No
Yes
Display in the appropriate portion of the video on display
No
Done?
Yes
End

FIG. 8

SYSTEM AND METHOD FOR DISTRIBUTING AND MANAGING DIGITAL VIDEO INFORMATION IN A VIDEO DISTRIBUTION NETWORK

CROSS-REFERENCE TO RELATED APPLICATIONS

This application is related to co-pending patent application Ser. No. 08/681,172, filed on Jun. 29, 1996, entitled "System and Method for Managing Digital Video Distribution Using Burst Transmission," by inventor Nathaniel Polish, which subject matter is hereby incorporated by reference. This related application has been commonly assigned to Instant Video Technology, Inc.

BACKGROUND OF THE INVENTION

1. Field of the Invention

This invention relates generally to digital video, and more particularly to a system and method for distributing and managing digital video information in a video distribution network.

2. Description of the Background Art

Conventional network digital video servers operate in a multithreaded environment to distribute digital video (and corresponding audio) information to multiple client computers. The client computers in turn offer VCR-like control of the digital video information including fast forward, rewind, pause, stop and play to a user. In some embodiments, the requesting client computer waits to receive the entire digital video work before beginning playback so that the user can enjoy seamless playback. However, to store an entire digital video work, significant local disk space and other memory resources are needed.

In other embodiments, the client computer requests appropriate portions of the digital video work and presents the portions on a display device as they are received. If the application software is properly designed and the network communications channel over which the client computer and the server are operating is predictable, then these requests may be totally transparent to the user and seamless playback is achieved. However, if the digital video data is not delivered within a rather tight time window, then the client computer stalls and playback is noticeably affected. At best, this can result in jerky playback of or noticeable pauses in the digital video work. At worst, the application software may not be designed to handle a network communication failure at all, which could result in various system failures more consequential than an annoyance in the digital video work playback.

One typical solution to this problem is to design the system for the worst-case scenario to prevent the problem from ever occurring. Accordingly, overall video server throughput is computed for a worst-case bit rate which, based on current digital video compression techniques, may be over five times the average required bit rate. The worst case scenario provides a system which inefficiently uses network bandwidth. Further, this approach does not function well in an environment, such as the internet, that does not have dedicated mechanisms for assuring available network bandwidth. The traditional methodology does not scale beyond a simple Local Area Network (LAN) environment. Therefore, a system and method are needed to distribute and manage digital video information in a video distribution network environment beyond a LAN.

SUMMARY OF THE INVENTION

The present invention provides a system and method for distributing and managing digital video information in a video distribution network. The video distribution network system includes client configuration data, a client video buffer for storing video information, a client video driver coupled to the client video buffer for presenting a portion of the video information on a display device, a current status manager for determining current client status information indicative of the portion of video information presented, a computations engine coupled to the client video buffer and to the current status manager for forwarding a burst of video information to the client video buffer based on the client configuration data and on the client status information, and a video buffer controller coupled to the client video buffer for controlling storage of the burst in the client video buffer.

The method includes the steps of storing video information in a client video buffer, controlling presentation of a first portion of the video information on a display device, forwarding to the computations engine client status information based on the presentation of the first portion of the video information, forwarding to the client video buffer a burst of video information based on client configuration data and on the client status information, and storing the burst of video information at locations in the client video buffer based on the client status information.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. **1** is a block diagram illustrating a burst video distribution network for managing digital video information in accordance with the present invention;

FIG. **2** is a block diagram illustrating the client computer for controlling the client system of FIG. **1**;

FIG. **3** is a block diagram illustrating the video server for controlling the video server system of FIG. **1**;

FIG. **4** is a block diagram illustrating the video buffer of FIG. **1** in an initial state;

FIG. **5** is a block diagram illustrating the video buffer of FIG. **1** after three seconds of play;

FIG. **6** is a block diagram illustrating the video buffer of FIG. **1** after two seconds of review;

FIG. **7** is a block diagram illustrating the video buffer of FIG. **1** after three seconds of play and two seconds of review; and

FIG. **8** is a flowchart illustrating a preferred method for managing digital video information in a vast video distribution network.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

FIG. **1** is a block diagram of a burst video distribution network **100**, which includes a video server system **105** coupled via a signal bus **110** to a client system **115**. The client system **115** is coupled via a communications channel **120** to an input device **125** and coupled via a communications channel **130** to a display device **135**. The client system **115** receives input control signals from a user via the input device **125** and accordingly forwards appropriate video information to a display device **135**.

The video server system **105** stores a video work **140** such as a music video, and includes a computations engine **145** for sizing and selecting a burst of video information from the video work **140** to forward to the client system **115**. The computations engine **145** selects and forwards the burst to a communications engine **150**, which compresses (e.g., using MPEG compression), packages, addresses and forwards the burst via the communications channel **110** to the client system **115**. Operations by the computations engine **145**,

including determining whether to allow a client system **115** to access the video server system **105** and selecting the most appropriate client system **115** in an environment having multiple client systems **115**, are described in greater detail in the cross-referenced patent application which is herein incorporated by reference.

The client system **115** includes a communications engine **155** for decompressing bursts from the video server system **105** and for compressing, packaging, addressing and transmitting information to the communications engine **150** as described herein. A client configuration engine **160** forwards client configuration data **162** to the communications engine **155**, which forwards the data **162** via the communications engine **150** to the computations engine **145**. The computations engine **145** then uses the data **162** to determine whether to allow the client system **115** to access the video server system **105**. That is, the computation engine **145** can examine the needs of the client system **115** to determine whether the video server system **105** has sufficient capacity available to grant access. If access is granted, the computations engine **145** then uses the client configuration data **162** to determine the size of the burst of video information to forward to the client system **115**.

Upon receipt of a burst of video information, the communications engine **155** forwards the burst to a video buffer **165** for storage. A video buffer controller **175** manages the video buffer **165**, i.e., tracks which portions of the video buffer **165** are available for storage. Management of the video buffer **165** is described in greater detail below with reference to FIGS. **4–7**. A video driver **170** extracts video information from the video buffer **165** to present on the display device **135**. After presenting the video information, the video driver **170** informs the video buffer controller **175** which updates the video buffer **165** accordingly.

A user operating an input device **125** can control the presentation of video information by effectively controlling the video driver **170**. That is, the user can select an input control signal from options including rewind **182**, fast forward, **184** play **186**, pause **188** and stop **190**. The input device **125** forwards the selected user control signal to an input interface **195**, which sends the user control signal to the video driver **170**. Based on the selected control signal, the video driver **170** regulates the video information extracted from the video buffer **165**. For example, if play **186** is selected, then the video driver **170** extracts video information from the video buffer **165** to produce a real-time video presentation. If rewind **182** is selected, then the video driver **170** selects relevant portions, such as every third frame, of the video information from the video buffer **165** in the reverse direction to produce a quick rewinding video presentation.

The video buffer controller **175** informs a current status manager **180** of the current status of video information stored in the video buffer **165**. The current status includes the amount of future video information and the amount of past video information. For example, the current status information may indicate that about two seconds of future video information (or about two megabytes of video information) and about eight seconds of past video information (or about eight megabytes of video information) remains in the video buffer **165**. Based on either the current status or at predetermined intervals, for example, every three seconds, the current status manager **180** forwards current status information to the computations engine **145** of the video server **105**. The computations engine **145** uses the current status information, as described in the cross-referenced patent application, to size and select a burst of video information for transmission to the client **115**.

Network **100** may further include a disk buffer system **195**, for use by a production manager for storing a copy of the video work **140**. As the video server system **105** forwards video information to the client system **115**, the disk buffer system **195** copies the video information to memory (not shown). The disk buffer system **195** includes most of the same components of client system **115**. However, the video buffer **165** is replaced by disk memory (not shown) which is sufficiently large to maintain the complete video work **140**. The video buffer controller **175** is replaced by a disk memory controller (not shown), which examines the video information and stores it chronologically. Alternatively, the disk buffer system **195** may be coupled within the client system **115** to the communications engine **155** and to the video driver **170** for storing a copy of the video work as presented on the display **135**.

FIG. **2** is a block diagram of a computer system **200** including a Central Processing Unit (CPU) **205** such as a Motorola Power PC® microprocessor or an Intel Pentium® microprocessor. An input port **210** for attaching an input device such as a keyboard and mouse, and an output port **215** for attaching an output device such as a Cathode Ray Tube (CRT) display are coupled via a signal bus **220** to the CPU **205**. A communications interface **225**, a data storage device **230**, such as Read Only Memory (ROM) and a magnetic disk, and a Random-Access Memory (RAM) **235** are further coupled via the signal bus **220** to the CPU **205**. The communications interface **250** is conventionally coupled via the communications channel **110** such as an internet to the communications interface **310** (FIG. **3**) of the video server **300**.

An operating system **240** includes a program that controls processing by the CPU **205**, and is typically stored in the data storage device **230** and loaded into the RAM **235** for execution. The client system **115** is a program which operates as described in FIG. **1**, and may be stored also in the data storage device **230** and loaded into the RAM **235** for execution by the CPU **205**.

FIG. **3** is a block diagram of a video server **300** including a CPU **305**, a communications interface **310**, a data storage device **320** and RAM **325** coupled to a signal bus **315**. The communications interface **310** is conventionally coupled via the communications channel **110** to the communications interface **225** of the client computer **200**.

An operating system **330** controls processing by the CPU **305**, and is typically stored in the data storage device **320** and loaded into the RAM **325** for execution. The video server system **105** is a program which operates as described in FIG. **1**, and may be stored also in the data storage device **320** and loaded into the RAM **325** for execution by the CPU **305**.

FIG. **4** is a block diagram illustrating the video buffer **165** in an initial state. The video buffer **165** is illustrated as including eleven sequential seconds of digital video information including the current second of video information (i.e., the second of video information to be played immediately on the display **135** based on the user's request via the input device **125**), five previous seconds of video information and five future seconds of video information. That is, the computations engine **145** of the video server system **105** determines, based on the client configuration data **162** and on the current status of the client system **115**, to send eleven seconds of the video work **140** to the client system **115**. The eleven seconds of information are stored in the video buffer **165** of the client system **115** and managed by the video buffer controller **175**.

The video buffer controller **175** maintains a pointer to the current second of video information (labeled as "t=0") and can easily compute positions of the other seconds of information relative to the position of the current pointer. Thus, the video buffer controller **175** currently knows that the five seconds of information preceding the current second of information [labeled as "(−5)," "(−4)," "(−3)," "(−2)" and "(−1)"] correspond to the five previous seconds of information and that the five seconds of information following the current second of information [labeled as "(+1)," "(+2)," "(+3)," "(+4)" and "(+5)"] correspond to the five future seconds of information. Although the video buffer controller **175** is illustrated and described as using only one pointer to the video buffer **165**, the video buffer controller **175** may use additional pointers to maintain the temporal organization of the video information without requiring that the physical organization correspond to the temporal organization.

FIG. **5** is a block diagram illustrating the video buffer **165** of FIG. **4** after three seconds of playback relative to the state of video buffer **165** in FIG. **4**. The video buffer controller **175** updates the pointer to reference the current second of video information which is three seconds of information beyond the current second discussed in FIG. **4**. Accordingly, the video buffer controller **175** determines that the video buffer **165** presently includes two future seconds and eight previous seconds of video information.

Since the video buffer controller **175** of the exemplary embodiment described herein maintains five future seconds and five previous seconds of information, the video buffer controller **175** determines that the three oldest seconds [labeled in FIG. **4** as "(−3)," "(−4)" and "(−5)"] are expendable. The current status manager **180** of the client system **115** sends current status information to the computations engine **145** of the video server system **105**, which accordingly sends three future seconds of information to the client system **115**. Upon receiving three future seconds of information, the video buffer controller **175** stores the three future seconds of information [labeled in FIG. **5** as "(+3)," "(+4)" and "(+5)"] in place of the three oldest seconds of information. The video buffer controller **175** can maintain the chronology of the eleven seconds of information by tracing all changes made to the video buffer **165**.

FIG. **6** is a block diagram illustrating the video buffer **165** of FIG. **4** after rewinding two seconds of information relative to the state of video buffer **165** in FIG. **4**. The video buffer controller **175** updates the current pointer to reference the current second of video information which is two seconds before the current second of information discussed in FIG. **4**. Accordingly, the video buffer controller **175** determines that the video buffer **165** includes three previous seconds of information and seven future seconds of information.

As stated above, the video buffer controller **175** of the embodiment described herein maintains five future seconds and five previous seconds of information. Thus, the video buffer controller **175** determines that the sixth future second [labeled in FIG. **4** as "(+4)"] and seventh future second of information [labeled in FIG. **4** as and "(+5)"] are expendable and will replace them upon receipt of the two previous seconds of information [labeled in FIG. **6** as "(−4)" and "(−5)"] from the video server system **105**.

FIG. **7** is a block diagram illustrating the video buffer **165** of FIG. **4** after playing three seconds of information and then rewinding two seconds of information. FIG. **5** illustrates the video buffer **165** after three seconds of playback. Thus, updating the video buffer **165** shown in FIG. **5** for the two seconds of rewind is appropriate.

The video buffer controller **175** updates the FIG. **5** current pointer to reference the current second of information which is two seconds of information into the past. Accordingly, the video buffer **165** includes seven future seconds of information and three previous seconds of information, and the video buffer controller **175** determines that the two most future seconds of information [labeled in FIG. **5** as "(+4)" and "(+5)"] are expendable. Thus, upon receipt of two previous seconds from the video server system **105**, the video buffer controller **175** will replace the two most future seconds of information with the two previous seconds of information [labeled in FIG. **7** as "(−4)" and "(−5)"].

FIG. **8** is a flowchart illustrating a preferred method **800** for managing digital video information in a video distribution network **100**. Method **800** begins with the video server system **105** in step **805** storing the complete video work **140** in memory **320** or **325**. The video work **140** may be stored to the video server system **105** by transferring it from a compact disk, from a live recording or via the internet from another computer (not shown).

In step **810**, video transfer capacity is negotiated as described in the co-pending, cross-referenced patent application. Briefly, the client system **115** requests access to the video work **140** and forwards client configuration information **1 62** to the video server system **105**. The computations engine **145** examines the client configuration information **162** to determine whether server system **105** has sufficient transfer capacity available to allow the connection. If sufficient capacity is unavailable, then the video server system **105** and the client system **115** either negotiate reduced configuration data or terminate the connection.

Assuming an acceptable video transfer capacity has been negotiated, the computations engine **145** in step **815** forwards a first burst of the video work **140** to the client system **1 15**, which stores the first burst in the video buffer **165**.

The client system **115** in step **820** polls for an input signal, such as play, **186**, rewind **182**, fast forward **184**, pause **188** and stop **190**. Based on the input signal, the video driver **170** in step **825** retrieves the appropriate video information, i.e., the current second of information, from the video buffer **165** and forwards it to the display **135** for presentation. For example, if play **186** is selected, then the video driver **170** forwards video information to the display **135** for presenting the images in real time. If fast-forward **184** is selected, then the video driver **170** retrieves selected portions of the video information, such as every third future frame, from the video buffer **165** and forwards them to the display **135** for simulating VCR-like fast-forward. If rewind **182** is selected, then the video driver **170** retrieves selected portions of the video information, such as every third previous frame, from the video buffer **165** and forwards them to the display **135** for simulating VCR-like rewind. If pause **188** is selected, then the video driver **170** retrieves the video information representing the current frame and forwards it to the display **135** for maintaining a VCR-like pause. Lastly, if stop **190** is selected, then the video driver **190** may retrieve predetermined "blue screen" video information and forward it to the display **135** for presentation. Other possible input signals include strobe, ½ play, double fast-forward, etc.

The video buffer controller **175** in step **830** receives from the video driver **170** display status information indicative of what has been presented on the display **135** and accordingly manages the video buffer **165**. The video buffer controller **175** maintains the status of the video buffer **165** and forwards the status to the current status manager **180**. The current status manager **180** in step **835** determines whether

predetermined criteria indicating that it is time to forward client status information to the computations engine **145** have been met. The predetermined criteria may be based on the number of seconds remaining in either direction. For example, if only three future seconds of video information remain in the video buffer **165**, then the current status manager **180** may determine that it is time to inform the computations engine **105** of the client status. If the predetermined criteria have been met, the current status manager **180** in step **840** forwards the current status information to the computations engine **145**. Method **800** then proceeds to step **845**. Otherwise, if the predetermined criteria have not been met, then method **800** jumps over step **840** and continues at step **845**.

In step **845**, the computations engine **145** determines whether server system predetermined criteria indicating time to forward more of the video work **140** to the client system **115** have been met. In this case, the server system predetermined criteria may be based on client configuration data **162** including the expected client video information consumption rate, the expected video information transfer latency, the current input signal selected, etc. If the predetermined criteria have been met, then the computations engine **145** in step **850** computes and forwards a burst of the video work **140** to the client system **115** and method **800** proceeds to step **855**. Otherwise, if the predetermined criteria have not been met, then the computations engine **145** jumps over step **850** and continues at step **855**. In step **855**, a determination is made whether to end the communication between the server system **105** and the client system **115**. If so, then method **800** ends. Otherwise, the method returns to step **820**.

The foregoing description of the preferred embodiments of the invention is by way of example only, and other variations of the above-described embodiments and methods are provided by the present invention. Components of this invention may be implemented using a programmed general purpose digital computer, using application specific integrated circuits, or using a network of interconnected conventional components and circuits. The embodiments described herein have been presented for purposes of illustration and are not intended to be exhaustive or limiting. The system is limited only by the following claims.

What is claimed is:

1. A computer-based method for managing digital video in a video distribution network, comprising the steps of:

receiving an input control signal for selecting from a video buffer a first portion of video information for output;

selecting for output, responsive to the input control signal, the first portion of the video information from the video buffer;

requesting a burst of video information from a video server, the requesting step including the step of forwarding client status information to the video server; and

replacing a second portion of the video information in the video buffer with the burst of video information.

2. The method of claim **1**, wherein the input control signal is a playback request.

3. The method of claim **1**, wherein the input control signal is a rewind request.

4. The method of claim **1**, wherein the input control signal is a fast-forward request.

5. The method of claim **1**, wherein the input control signal is a pause request.

6. The method of claim **1**, wherein the input control signal is a stop request.

7. The method of claim **1**, wherein the client status information indicates the amount of future video information and the amount of previous video information.

8. The method of claim **1**, further comprising, after requesting the burst, the step of receiving from the video server the burst of video information responsive to the client status information.

9. The method of claim **8**, further comprising the step of selecting the second portion based on the client status information.

10. The method of claim **9**, wherein the step of selecting includes selecting the most expendable locations in the video buffer as the second portion.

11. A computer-based method, comprising the steps of:

storing video information in a video buffer;

receiving an input control signal;

presenting a first portion of the video information on a display device responsive to the input control signal;

forwarding current status information, indicative of the first portion presented, to a video server system;

receiving a burst of video information, responsive to the current status information, from the video server; and

replacing in the video buffer a second portion of video information, based on the current status information, with the burst of video information.

12. The method of claim **1**, wherein the video buffer is sized to store only a predetermined amount of video information.

13. The method of claim **11**, wherein the input control signal is a playback request.

14. The method of claim **11**, wherein the input control signal is a rewind request.

15. The method of claim **11**, wherein the input control signal is a fast-forward request.

16. The method of claim **11**, wherein the input control signal is a pause request.

17. The method of claim **11**, wherein the input control signal is a stop request.

18. The method of claim **11**, wherein the current status information indicates the amount of future video information and the amount of previous video information.

19. The method of claim **11**, further comprising, after receiving the burst, the step of selecting the most expendable locations in the video buffer as the second portion.

20. A system, comprising:

a video buffer for storing video information;

a video driver coupled to the video buffer for presenting a first portion of the video information on a display device;

an input interface coupled to the video driver for receiving an input control signal which controls the video driver;

a current status manager for forwarding current client status information, indicative of the first portion presented, to a video server system;

a communications engine coupled to the video buffer for receiving a burst of video information from the video server system; and

a video buffer controller coupled to the video buffer for controlling storage of the burst in the video buffer based on the current client status information.

21. The system of claim **20**, wherein the video buffer is sized to store only a predetermined amount of video information.

22. The system of claim **20**, wherein the input control signal is a playback request.

23. The system of claim **20**, wherein the input control signal is a rewind request.

24. The system of claim **20**, wherein the input control signal is a fast-forward request.

25. The system of claim **20**, wherein the input control signal is a pause request.

26. The system of claim **20**, wherein the input control signal is a stop request.

27. The system of claim **20**, wherein the current status information indicates the amount of future video information and the amount of previous video information.

28. The system of claim **27**, wherein the current status information further indicates the input control signal.

29. The system of claim **20**, wherein the video buffer controller stores the burst in the most expendable locations in the video buffer.

30. A computer-readable storage medium storing program code for causing a computer to perform the steps of:

storing video information in a video buffer;

receiving an input control signal;

presenting a first portion of the video information on a display device responsive to the input control signal;

forwarding current status information, indicative of the first portion presented, to a video server system;

receiving a burst of video information, responsive to the current status information, from the video server; and

replacing in the video buffer a second portion of video information, based on the current status information, with the burst of video information.

31. A system, comprising:

means for storing video information;

means for receiving an input control signal;

means for presenting a first portion of the video information on a display device responsive to the input control signal;

means for forwarding current status information, indicative of the first portion presented, to a video server system;

means for receiving a burst of video information, responsive to the current status information, from the video server; and

means for replacing in the video buffer a second portion of video information, based on the current status information, with the burst of video information.

32. A computer-based method, comprising the steps of:

storing video information in client video buffer;

controlling presentation of a first portion of the video information on a display device;

forwarding to a computations engine client status information based on the presentation of the first portion of the video information;

forwarding to the client video buffer a burst of video information based on client configuration data and on the client status information; and

storing the burst of video information at locations in the client video buffer based on the client status information.

33. The method of claim **32**, wherein the client video buffer is sized to store only a predetermined amount of video information, and the client configuration data includes a value representing the predetermined amount.

34. The method of claim **32**, wherein the client status information indicates the amount of future video information and the amount of previous video information.

35. The method of claim **32**, wherein storing the burst includes the step of storing the burst at the most expendable locations in the video buffer.

36. The method of claim **32**, further comprising, before controlling presentation of the first portion, the step of receiving an input control signal and wherein the step of controlling presentation is responsive to the input control signal.

37. The method of claim **36**, wherein the input control signal is a playback request.

38. The method of claim **36**, wherein the input control signal is a rewind request.

39. The method of claim **36**, wherein the input control signal is a fast-forward request.

40. The method of claim **36**, wherein the input control signal is a pause request.

41. The method of claim **36**, wherein the input control signal is a stop request.

42. A network system, comprising:

client configuration data;

a client video buffer for storing video information;

a client video driver coupled to the client video buffer for presenting a portion of the video information on a display device;

a current status manager for determining current client status information indicative of the portion of video information presented;

a computations engine coupled to the client video buffer and to the current status manager for forwarding a burst of video information to the client video buffer based on the client configuration data and on the client status information; and

a video buffer controller coupled to the client video buffer for controlling storage of the burst in the client video buffer.

43. The system of claim **42**, wherein the client configuration data includes a value indicating the size of the client video buffer.

44. The system of claim **42**, wherein the client status information indicates the amount of future video information and the amount of previous video information.

45. The system of claim **42**, wherein the step of storing includes storing the burst of video information at the locations in the client video buffer having the most expendable data.

46. The system of claim **42**, further comprising an input interface for receiving from a user an input control signal which controls the presentation of the first portion.

47. The system of claim **46**, wherein the current client status information includes the input control signal.

* * * * *

EXHIBIT F

US006850965B2

(12) **United States Patent**
Allen

(10) **Patent No.: US 6,850,965 B2**
(45) **Date of Patent: Feb. 1, 2005**

(54) **METHOD FOR CONNECTION ACCEPTANCE AND RAPID DETERMINATION OF OPTIMAL MULTI-MEDIA CONTENT DELIVERY OVER NETWORK**

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(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 603 days.

(21) Appl. No.: **09/893,364**

(22) Filed: **Jun. 26, 2001**

(65) **Prior Publication Data**

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Related U.S. Application Data

(63) Continuation-in-part of application No. 09/344,688, filed on Jun. 25, 1999

(60) Provisional application No. 60/108,777, filed on Nov. 17, 1998.

(51) **Int. Cl.**[7] **G06F 15/16**

(52) **U.S. Cl.** **709/203**; 709/226; 709/229; 709/232; 709/233; 709/234; 370/237; 370/389; 370/463

(58) **Field of Search** 709/203, 226, 709/233, 234; 370/232, 239, 389, 468

(56) **References Cited**

U.S. PATENT DOCUMENTS

5,583,995 A 12/1996 Gardner et al.

5,982,748 A * 11/1999 Yin et al. 370/232

(List continued on next page.)

OTHER PUBLICATIONS

Kim, S. et al., "Call Admission Control for Prioritized Adaptive Multimedia Services in Wireless/Mobil Networks", *VTC 2000–Spring. 2000 IEEE 51st . Vehicular Technology Conference Proceedings. Tokyo, Japan, May 15–18, 2000, IEEE Vehicular Technology Conference, New York, NY.,* (May 15, 2000) IEEE, US, vol. 2 of 3: Conf. 51, pp. 1536–1540.

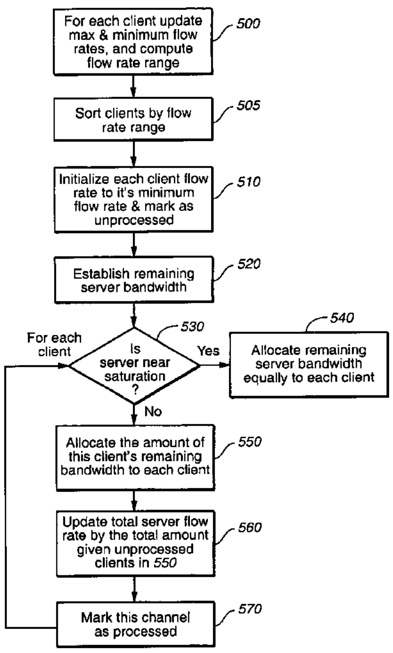
(List continued on next page.)

Primary Examiner—Hosain Alam
Assistant Examiner—ThuHa Nguyen
(74) *Attorney, Agent, or Firm*—Burns, Doane, Swecker & Mathis, L.L.P.

(57) **ABSTRACT**

A method of bandwidth allocation for delivery of stored digital content from at least one server device to at least one client device by way of a network is disclosed. The method begins by prescribing a control variable which represents a target flow rate from the server device to each client device. Next, time-varying constraints on the flow rate of the content are determined. A cost function of the control variable for each client is determined. The cost function corresponds to a maximized value of the control variable. Finally, bandwidth is prescribed to each client based upon the value of the control variable maximized by the cost function. In this respect, the method achieves optimal allocation of bandwidth between the server and the respective clients.

33 Claims, 15 Drawing Sheets



U.S. PATENT DOCUMENTS

5,995,488	A	*	11/1999	Kalkunte et al.	370/232
6,047,328	A	*	4/2000	Charny et al.	709/233
6,052,384	A	*	4/2000	Huang et al.	370/468
6,069,894	A		5/2000	Holender et al.	
6,125,396	A	*	9/2000	Lowe	709/234
6,192,029	B1	*	2/2001	Averbuch et al.	370/229
6,212,169	B1		4/2001	Bawa et al.	
6,240,103	B1	*	5/2001	Schoenblum et al.	370/468
6,295,294	B1	*	9/2001	Odlyzko	370/389
6,331,986	B1	*	12/2001	Mitra et al.	370/468
6,400,686	B1	*	6/2002	Ghanwani et al.	370/232
6,493,317	B1	*	12/2002	Ma	370/237
6,597,662	B1	*	7/2003	Kumar et al.	370/236
6,647,419	B1	*	11/2003	Mogul	709/226
2003/0016664	A1	*	1/2003	MeLampy et al.	370/389
2003/0072327	A1	*	4/2003	Fodor et al.	370/468

OTHER PUBLICATIONS

Reininger, D. et al., "A Dynamic Quality of Service Framework for Video in Broadband Networks", *IEEE Networks, IEEE Inc., New York, US* (Nov. 6, 1998), 12:6, pp 22–34.
Beard, C. et al., "Connention Admission Control for Differentiating Priority Traffic on Public Networks", *Military Communications Conference Proceedings, 1999, Milcom 1999. IEEE Atlantic Cith, NJ, USA Oct. 31–Nov. 3, 1999, Piscataway, NJ, USA, IEEE, US* (Oct. 31, 1999), pp. 1401–1408.

* cited by examiner

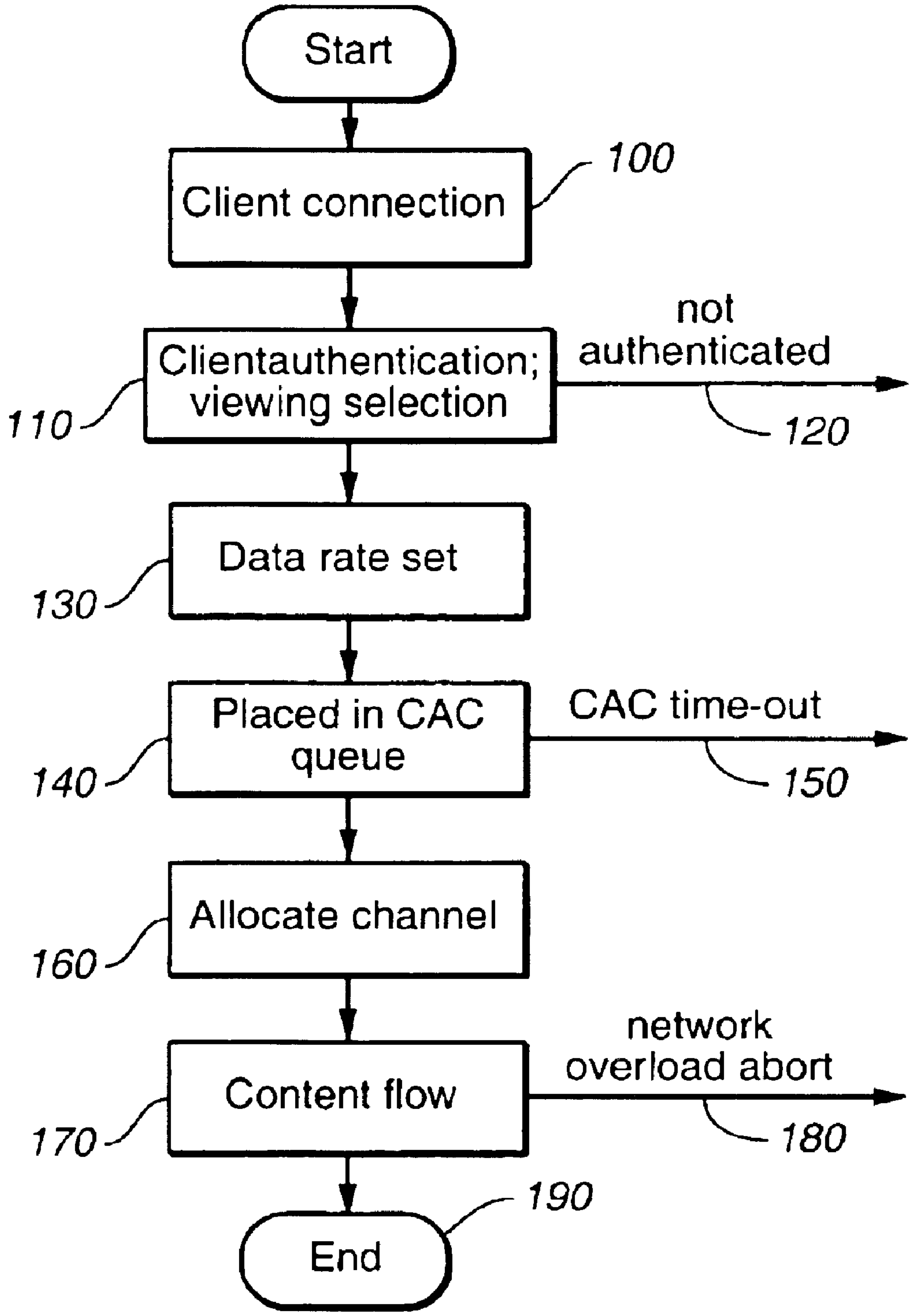

Start
Client connection
100
not authenticated
Clientauthentication; viewing selection
110
120
Data rate set
130
Placed in CAC queue
CAC time-out
140
150
Allocate channel
160
network overload abort
Content flow
170
180
190
End

FIG. 1

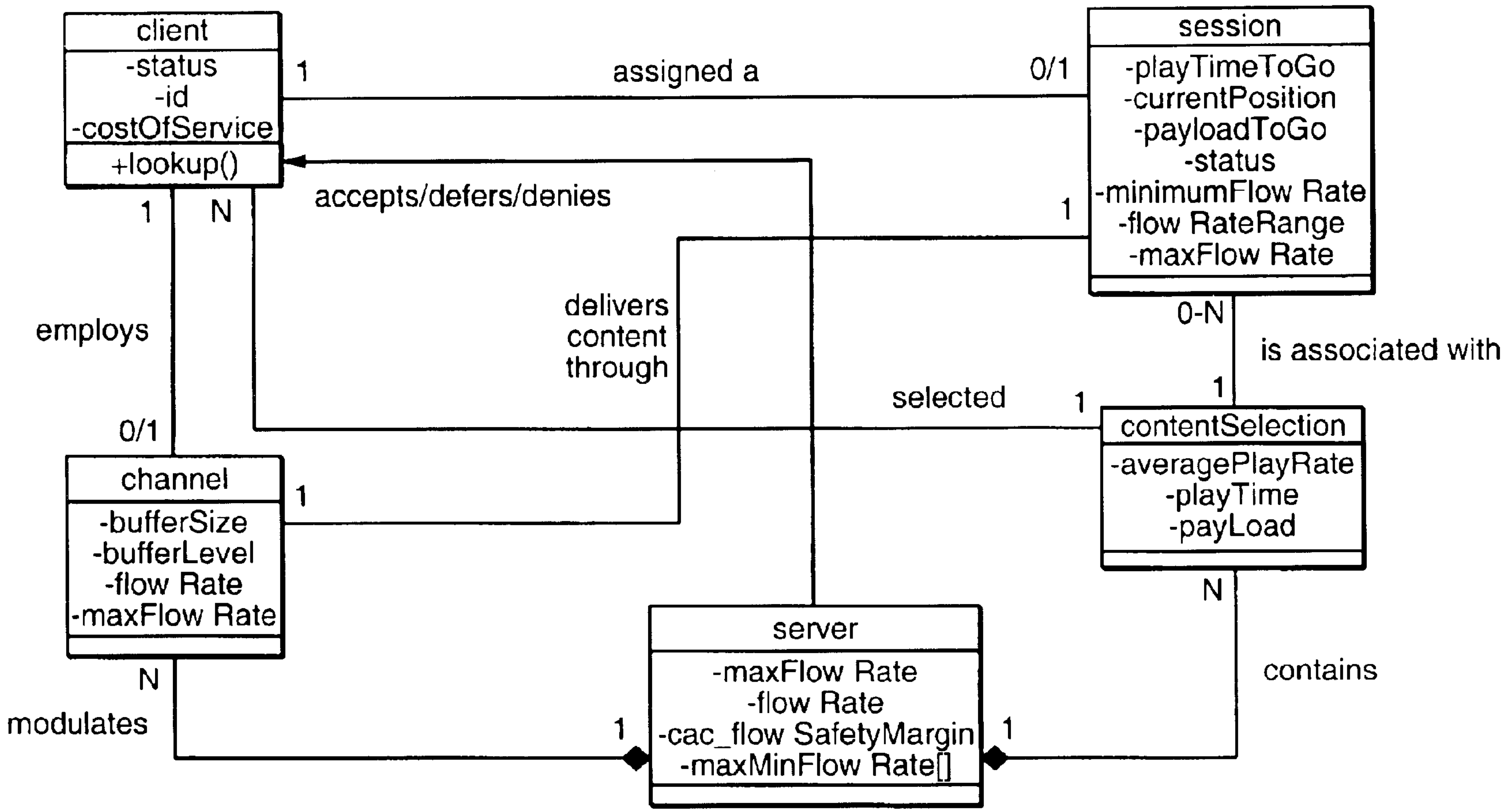

client
-status
-id
-costOfService
+lookup()
session
-playTimeToGo
-currentPosition
-payloadToGo
-status
-minimumFlow Rate
-flow RateRange
-maxFlow Rate
assigned a
1
0/1
accepts/defers/denies
1
N
1
employs
0/1
delivers
content
through
selected
1
0-N
is associated with
1
contentSelection
-averagePlayRate
-playTime
-payLoad
channel
-bufferSize
-bufferLevel
-flow Rate
-maxFlow Rate
1
N
modulates
server
-maxFlow Rate
-flow Rate
-cac_flow SafetyMargin
-maxMinFlow Rate[]
1
1
N
contains

FIG._2

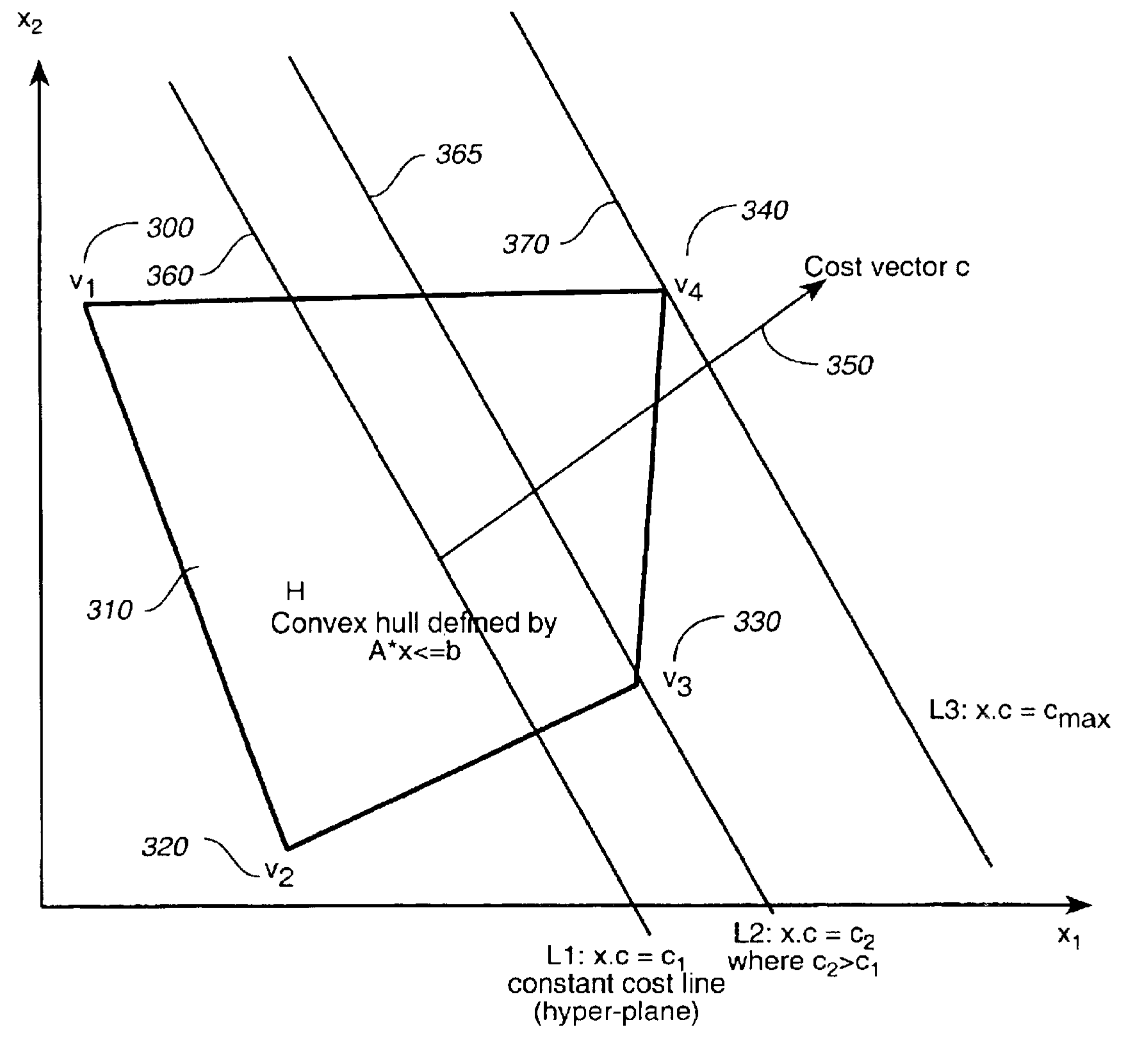
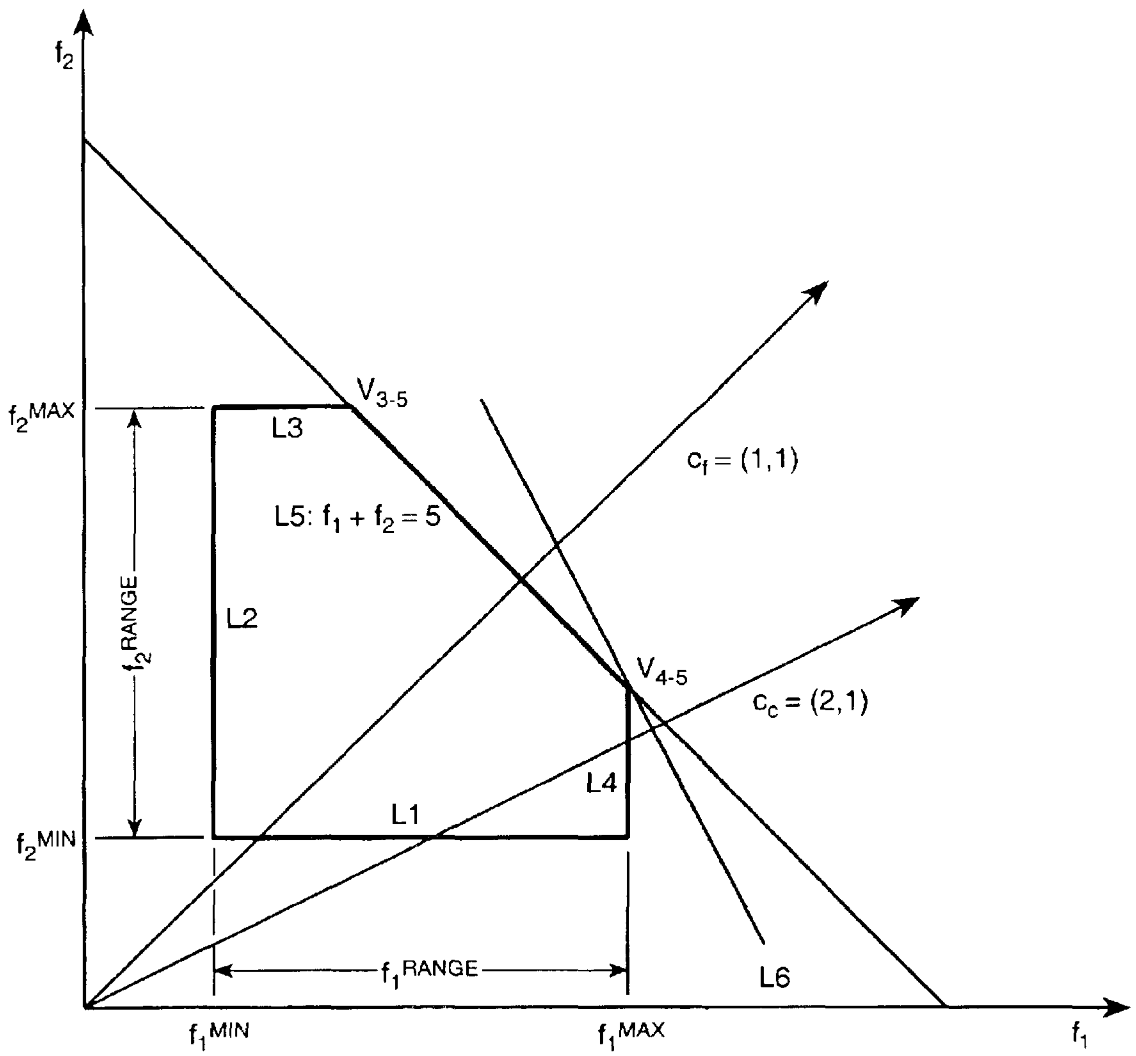
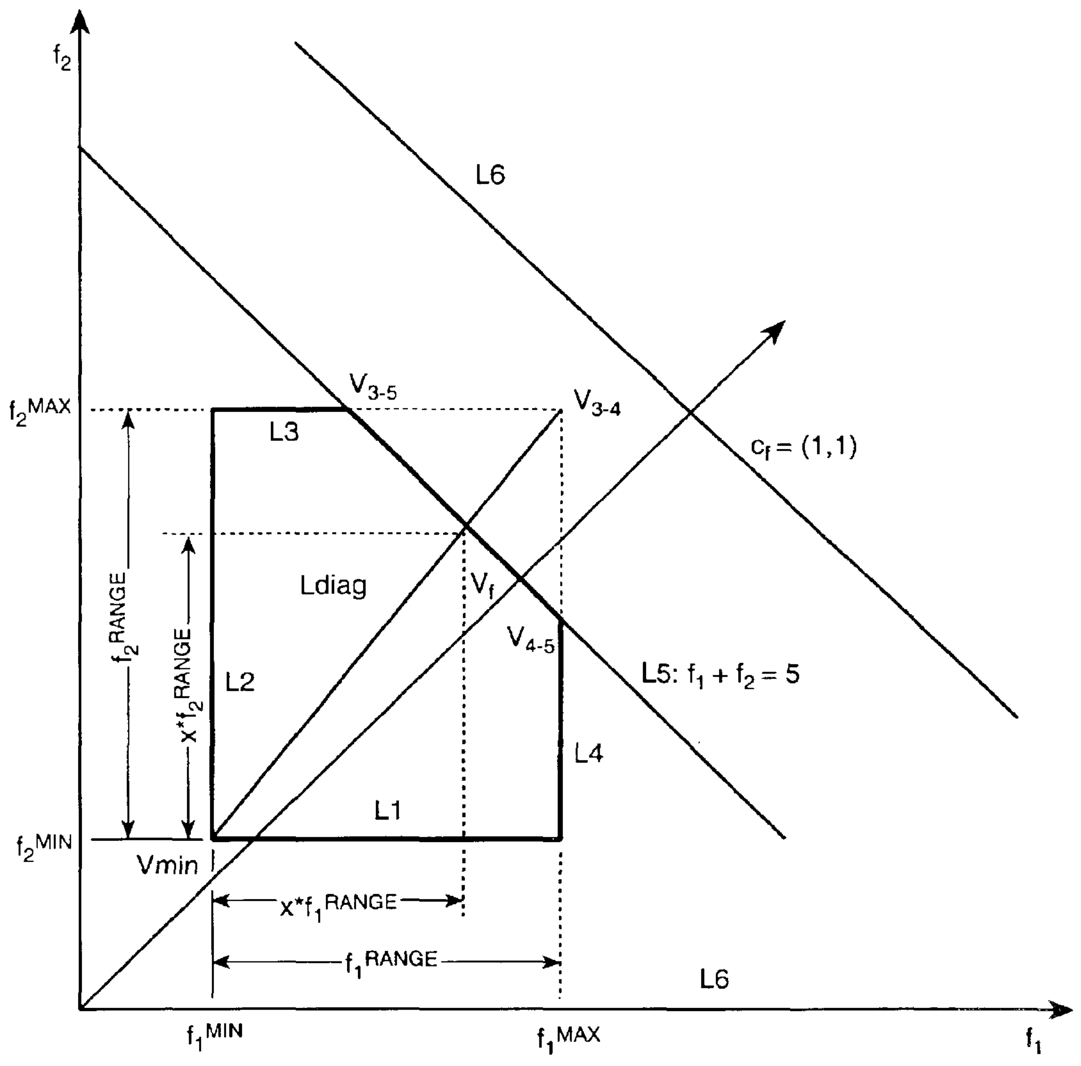

x_2
365
300
370
340
v_1
360
v_4
Cost vector c
350
310
H
Convex hull defined by
A*x<=b
330
v_3
L3: $x.c = c_{max}$
320
v_2
x_1
L1: $x.c = c_1$
constant cost line
(hyper-plane)
L2: $x.c = c_2$
where $c_2 > c_1$

FIG._3



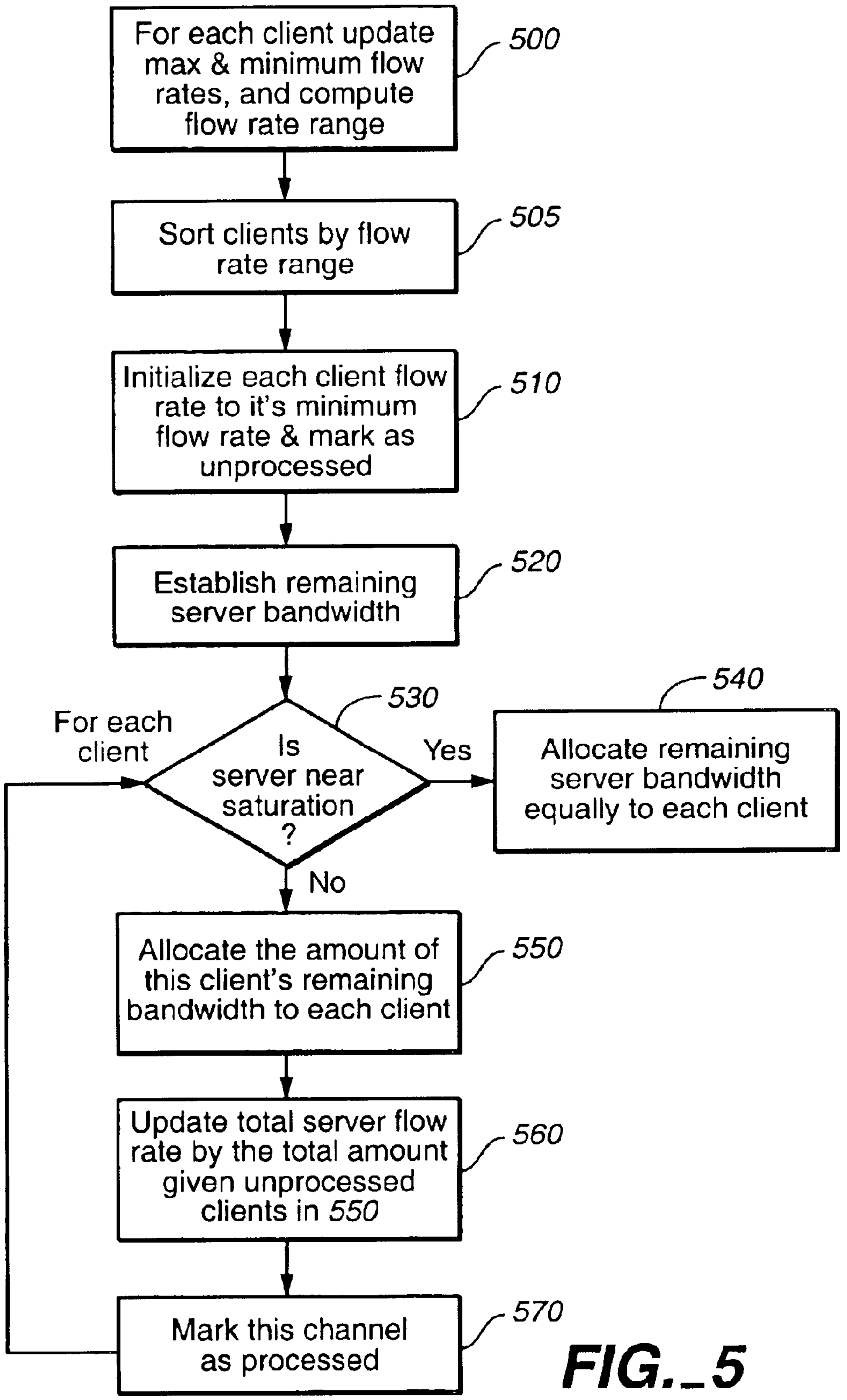
f_2
f_2MAX
f_2MIN
f_2RANGE
V_{3-5}
L3
L5: $f_1 + f_2 = 5$
L2
V_{4-5}
L4
L1
L6
$c_f = (1,1)$
$c_c = (2,1)$
f_1RANGE
f_1MIN
f_1MAX
f_1

FIG._4a



f_2
L6
f_2MAX
V_{3-5}
V_{3-4}
L3
$c_f = (1,1)$
Ldiag
V_f
f_2RANGE
x^*f_2RANGE
V_{4-5}
L2
L5: $f_1 + f_2 = 5$
L4
L1
f_2MIN
Vmin
x^*f_1RANGE
f_1RANGE
L6
f_1MIN
f_1MAX
f_1

FIG._4b



For each client update max & minimum flow rates, and compute flow rate range
500
Sort clients by flow rate range
505
Initialize each client flow rate to it's minimum flow rate & mark as unprocessed
510
Establish remaining server bandwidth
520
For each client
Is server near saturation ?
530
Yes
Allocate remaining server bandwidth equally to each client
540
No
Allocate the amount of this client's remaining bandwidth to each client
550
Update total server flow rate by the total amount given unprocessed clients in 550
560
Mark this channel as processed
570

FIG._5

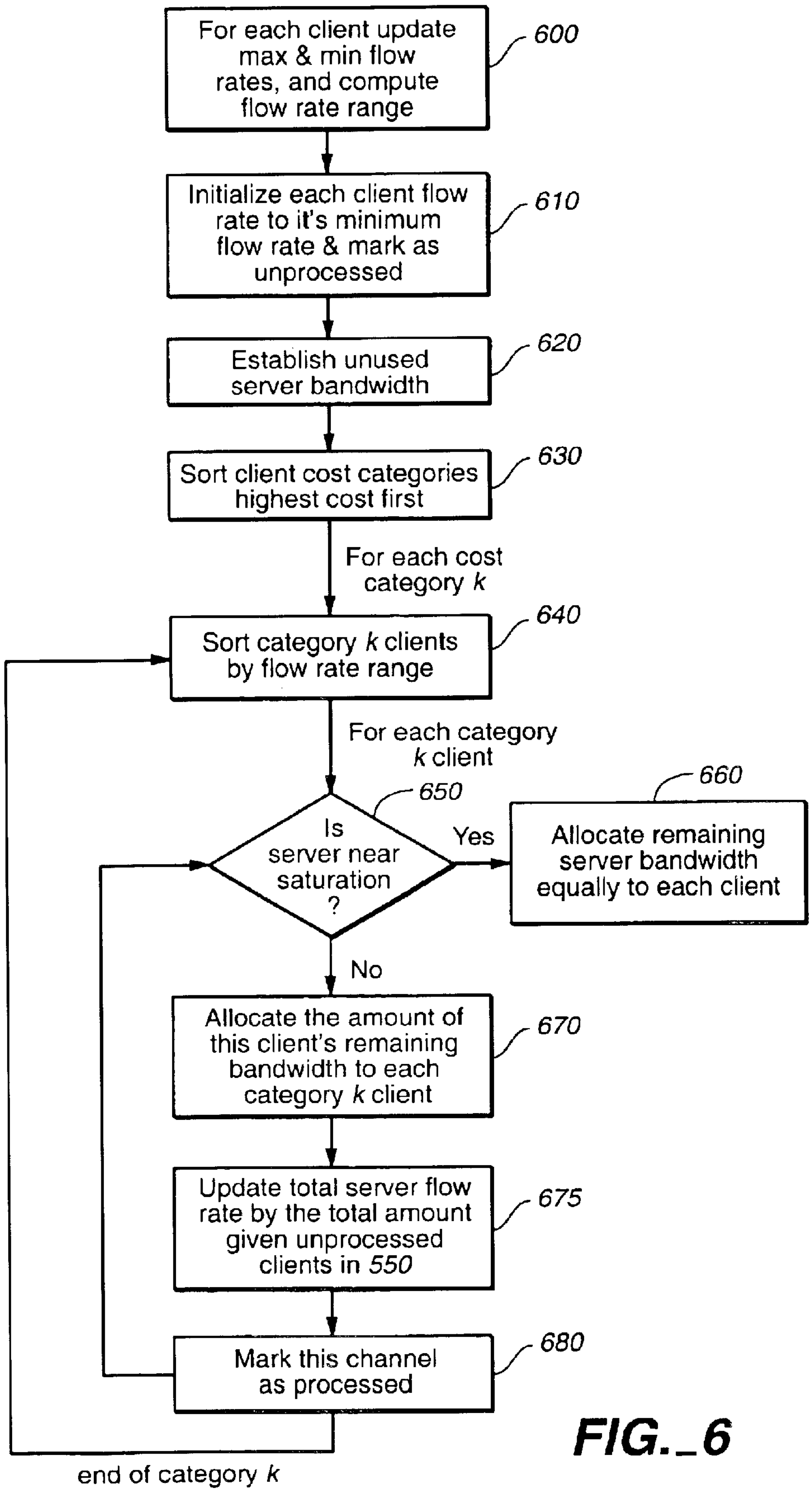

For each client update max & min flow rates, and compute flow rate range
600
Initialize each client flow rate to it's minimum flow rate & mark as unprocessed
610
Establish unused server bandwidth
620
Sort client cost categories highest cost first
630
For each cost category k
Sort category k clients by flow rate range
640
For each category k client
650
Is server near saturation ?
Yes
Allocate remaining server bandwidth equally to each client
660
No
Allocate the amount of this client's remaining bandwidth to each category k client
670
Update total server flow rate by the total amount given unprocessed clients in 550
675
Mark this channel as processed
680
end of category k

FIG._6

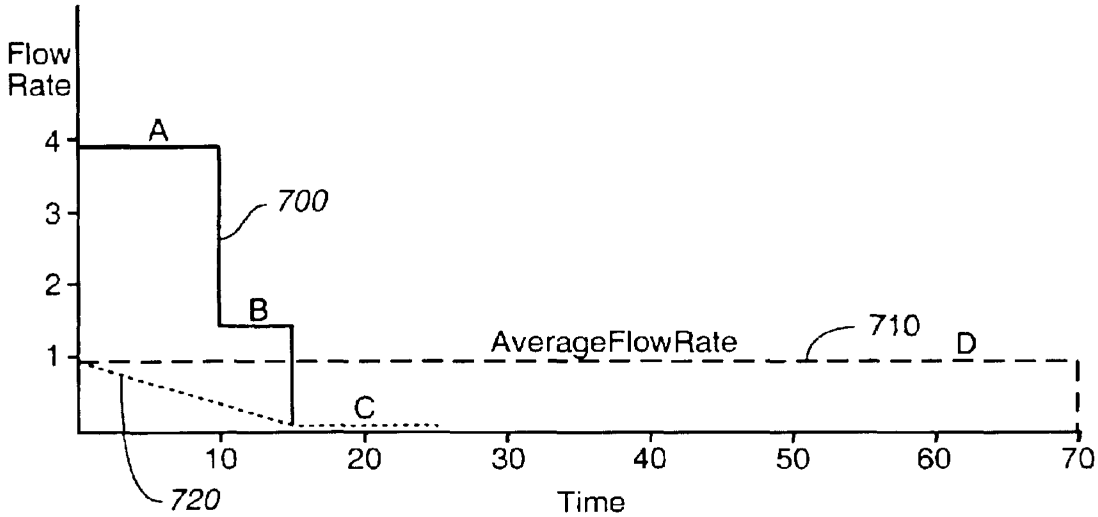

Flow Rate
4
3
2
1
A
700
B
AverageFlowRate
710
D
C
720
10
20
30
40
50
60
70
Time

FIG._7

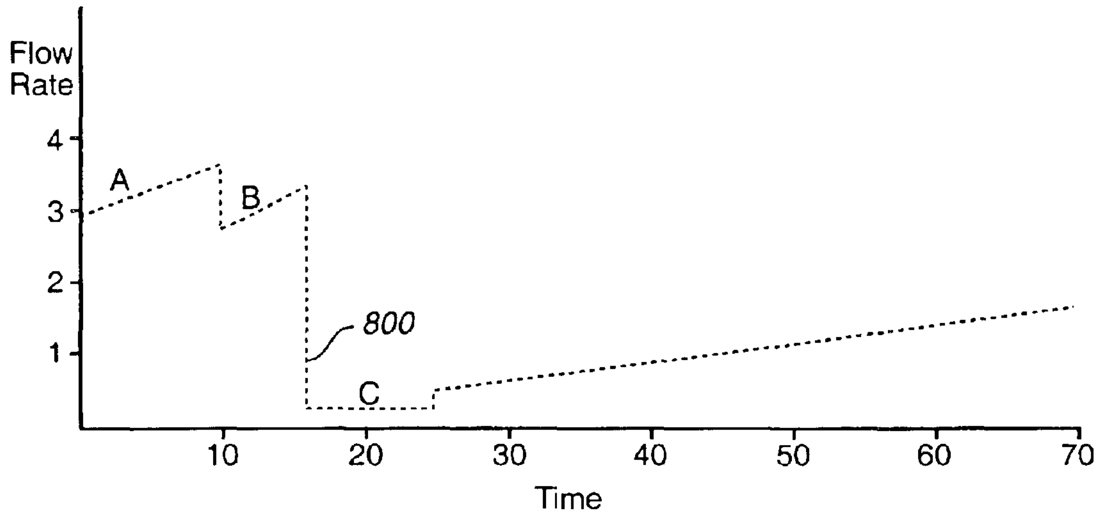

Flow Rate
4
3
2
1
A
B
800
C
10
20
30
40
50
60
70
Time

FIG._8

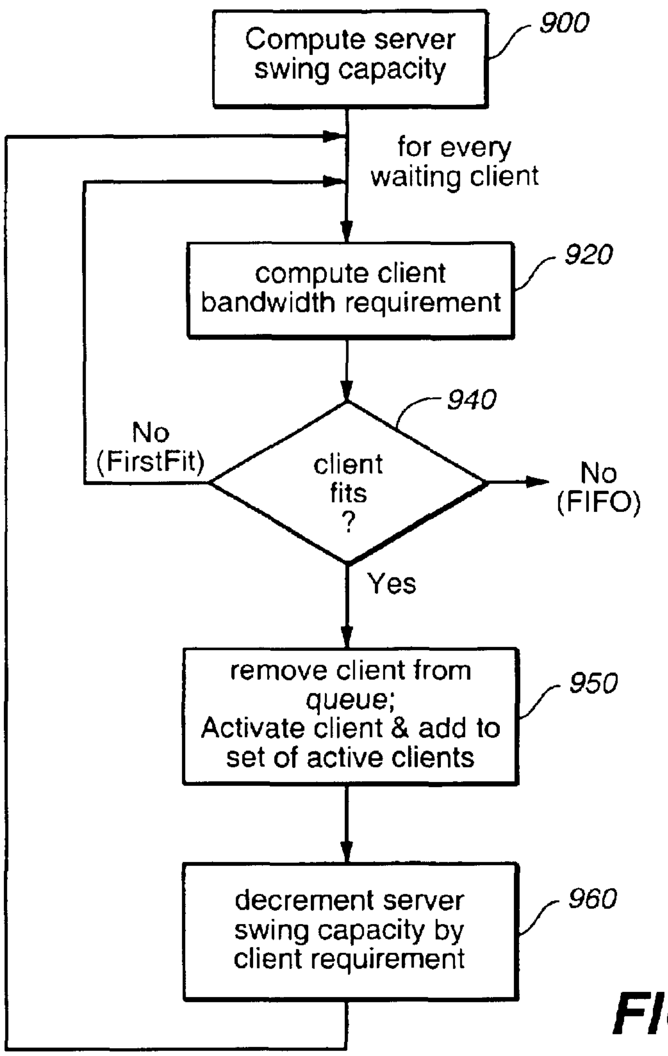

Compute server swing capacity
900
for every waiting client
compute client bandwidth requirement
920
940
No (FirstFit)
client fits ?
No (FIFO)
Yes
remove client from queue; Activate client & add to set of active clients
950
decrement server swing capacity by client requirement
960

FIG._9

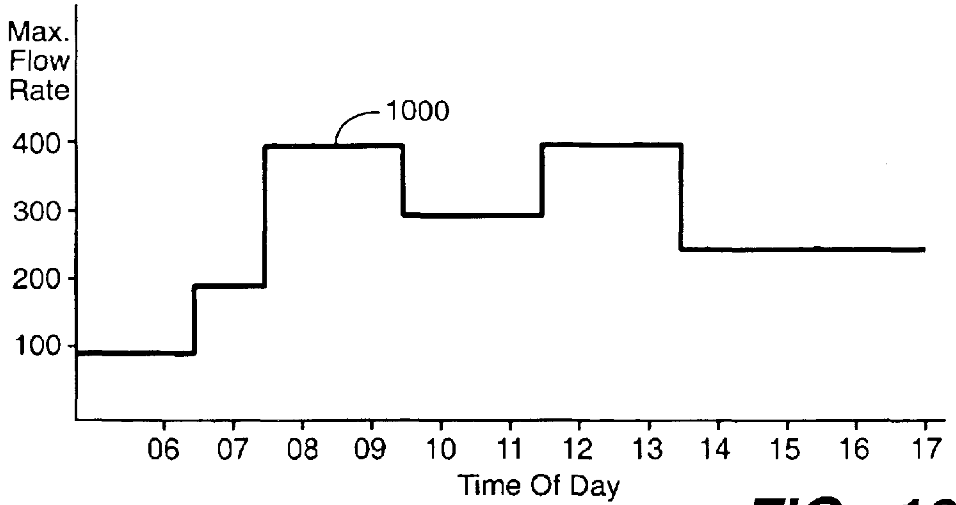
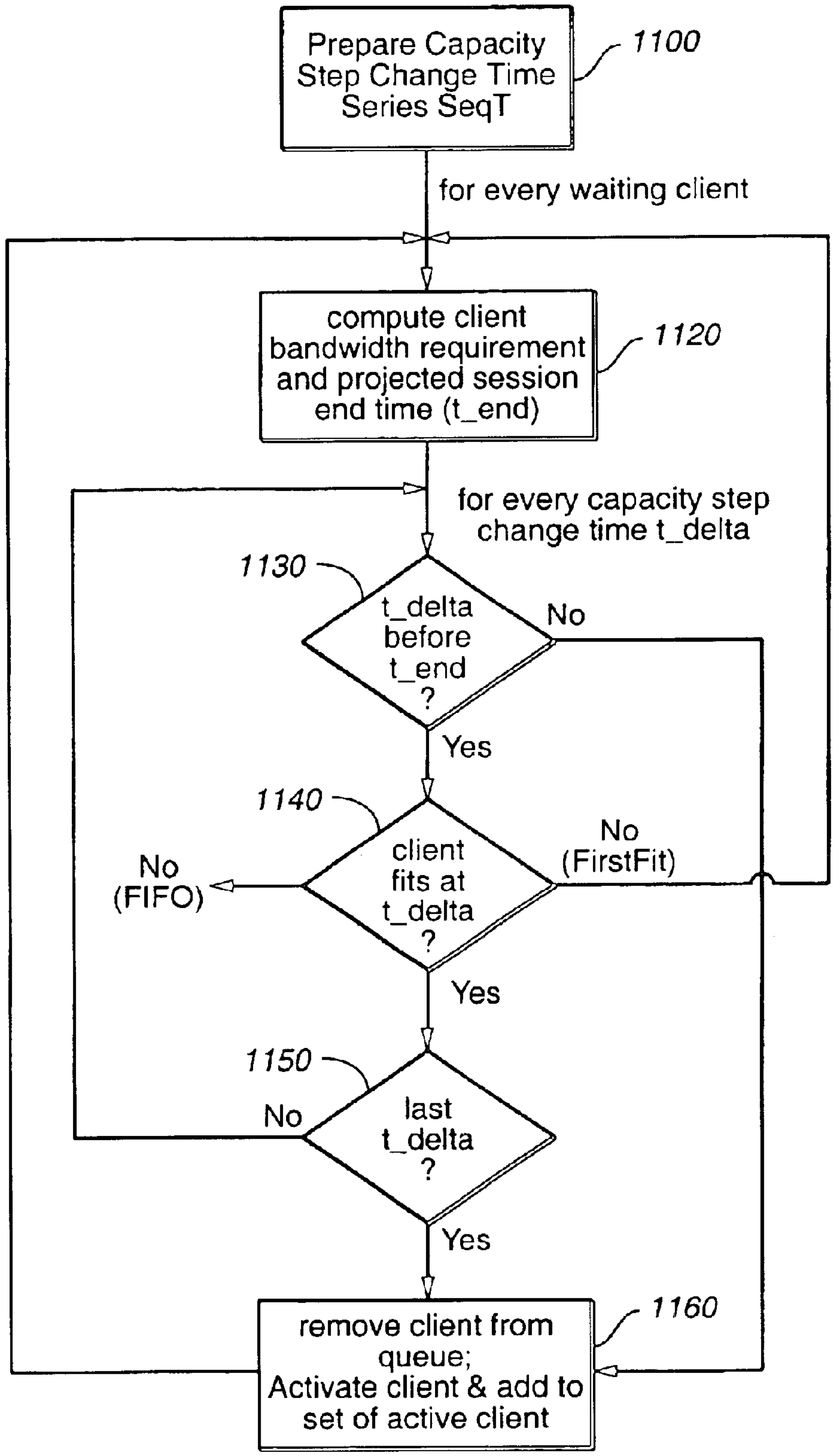
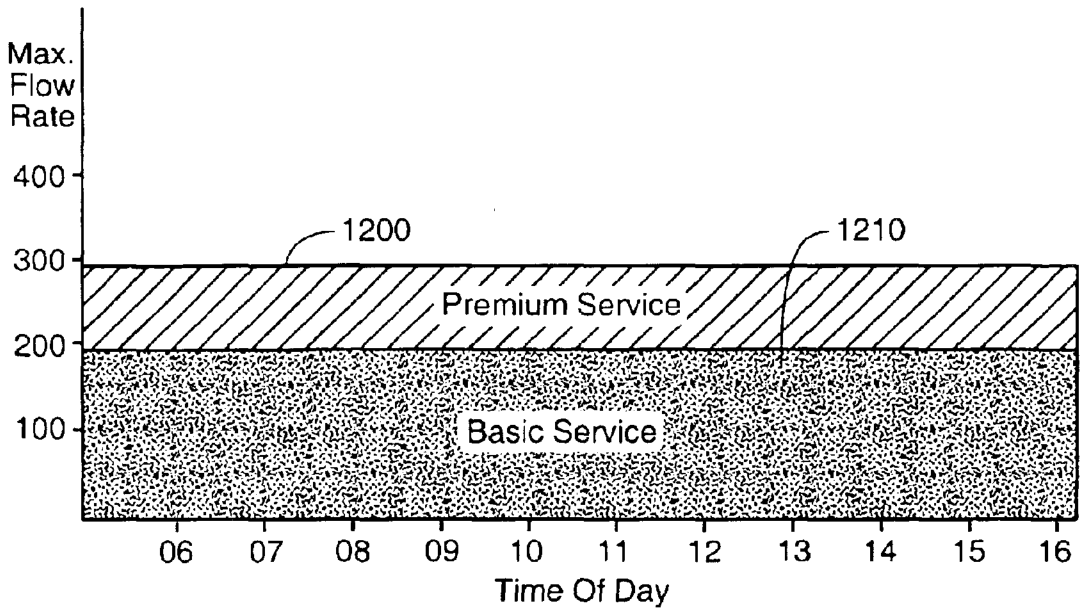

Max. Flow Rate
400
300
200
100
1000
06
07
08
09
10
11
12
13
14
15
16
17
Time Of Day

FIG._10



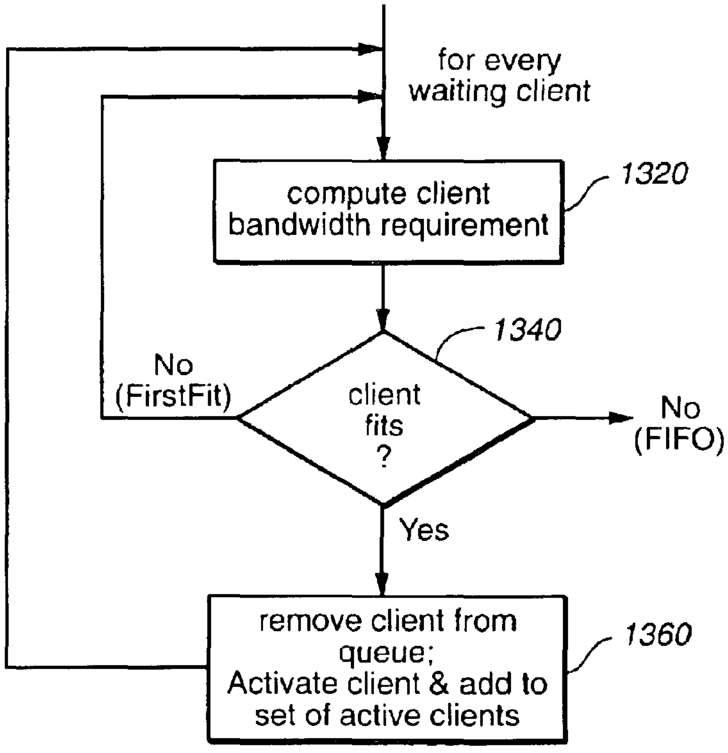
Prepare Capacity Step Change Time Series SeqT
1100
for every waiting client
compute client bandwidth requirement and projected session end time (t_end)
1120
for every capacity step change time t_delta
1130
t_delta before t_end ?
No
Yes
1140
client fits at t_delta ?
No (FIFO)
No (FirstFit)
Yes
1150
last t_delta ?
No
Yes
remove client from queue; Activate client & add to set of active client
1160

FIG._11



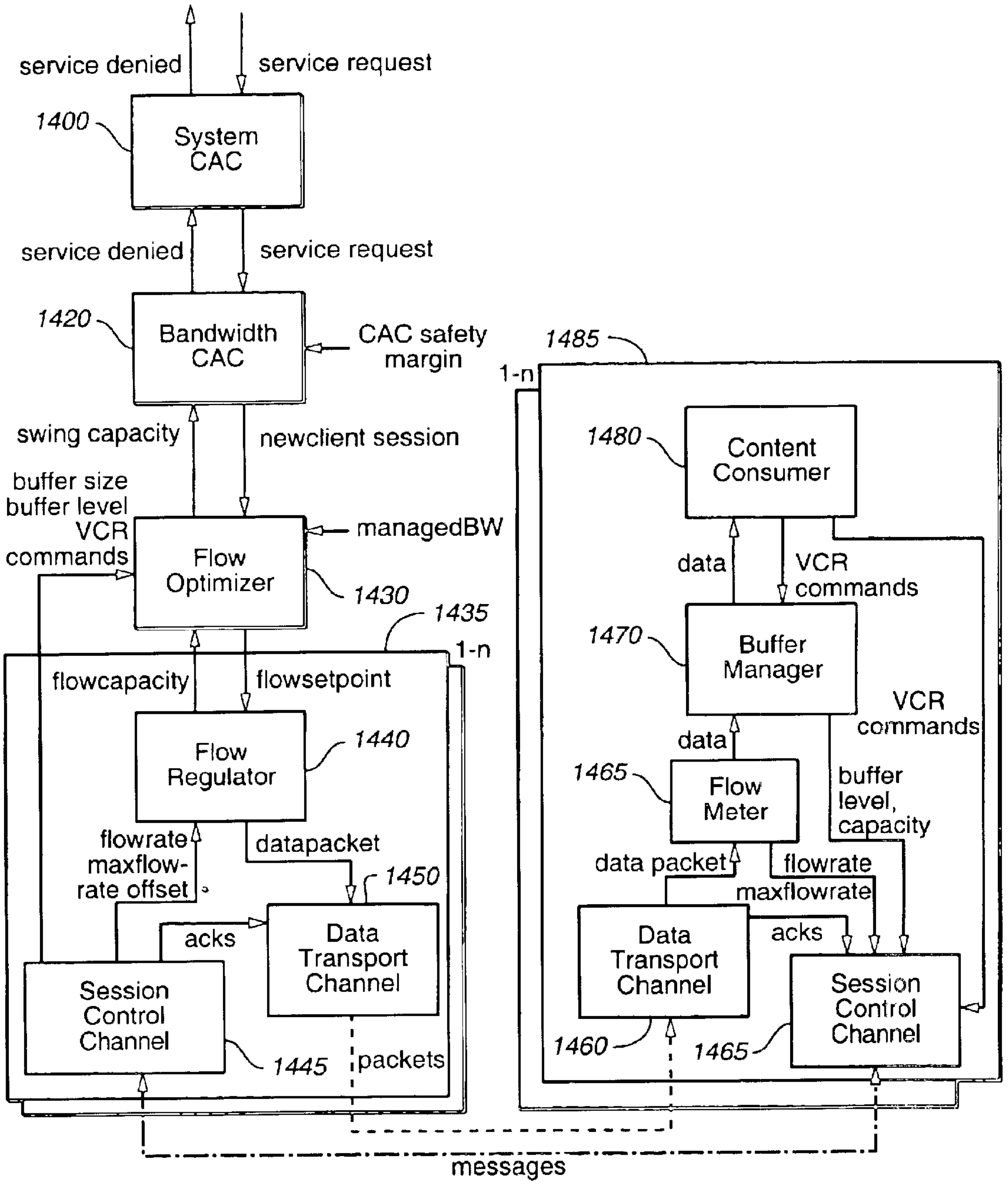
Max. Flow Rate
400
300
200
100
1200
1210
Premium Service
Basic Service
06
07
08
09
10
11
12
13
14
15
16
Time Of Day

FIG._12



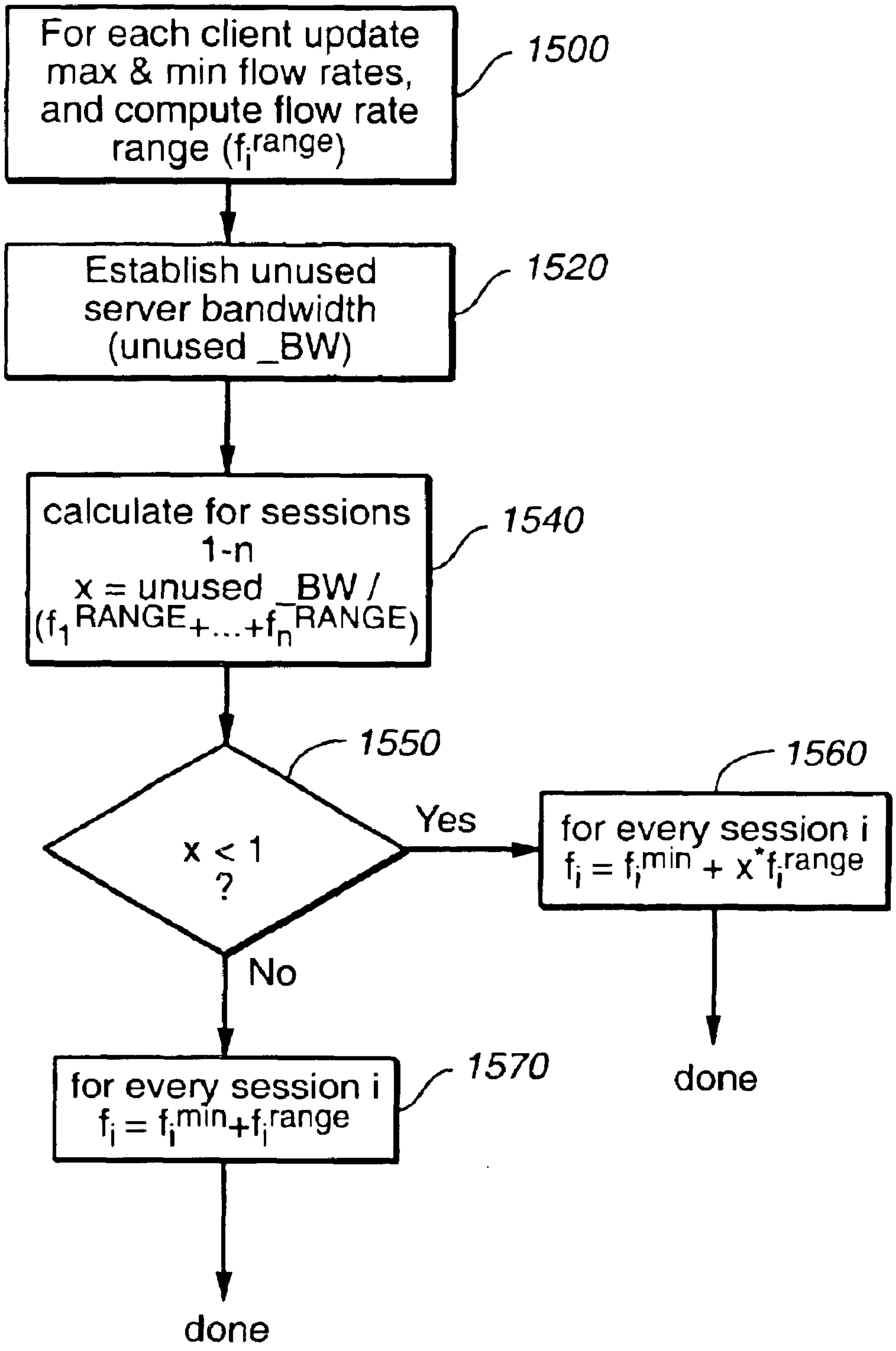
for every waiting client
compute client bandwidth requirement
1320
1340
client fits ?
No (FirstFit)
No (FIFO)
Yes
remove client from queue; Activate client & add to set of active clients
1360

FIG._13



service denied
service request
1400
System CAC
service denied
service request
1420
Bandwidth CAC
CAC safety margin
swing capacity
newclient session
buffer size
buffer level
VCR commands
Flow Optimizer
managedBW
1430
1435
1-n
flowcapacity
flowsetpoint
Flow Regulator
1440
flowrate
maxflow-rate offset
datapacket
1450
acks
Data Transport Channel
Session Control Channel
1445
packets
1485
1-n
1480
Content Consumer
data
VCR commands
1470
Buffer Manager
VCR commands
data
1465
Flow Meter
buffer level, capacity
data packet
flowrate
maxflowrate
acks
Data Transport Channel
Session Control Channel
1460
1465
messages

FIG._14


For each client update max & min flow rates, and compute flow rate range (f_i^{range})
1500
Establish unused server bandwidth (unused _BW)
1520
calculate for sessions 1-n
x = unused _BW / (f_1^{RANGE}+...+f_n^{RANGE})
1540
1550
x < 1 ?
Yes
No
1560
for every session i
$f_i = f_i^{min} + x^* f_i^{range}$
done
1570
for every session i
$f_i = f_i^{min} + f_i^{range}$
done

I ._15

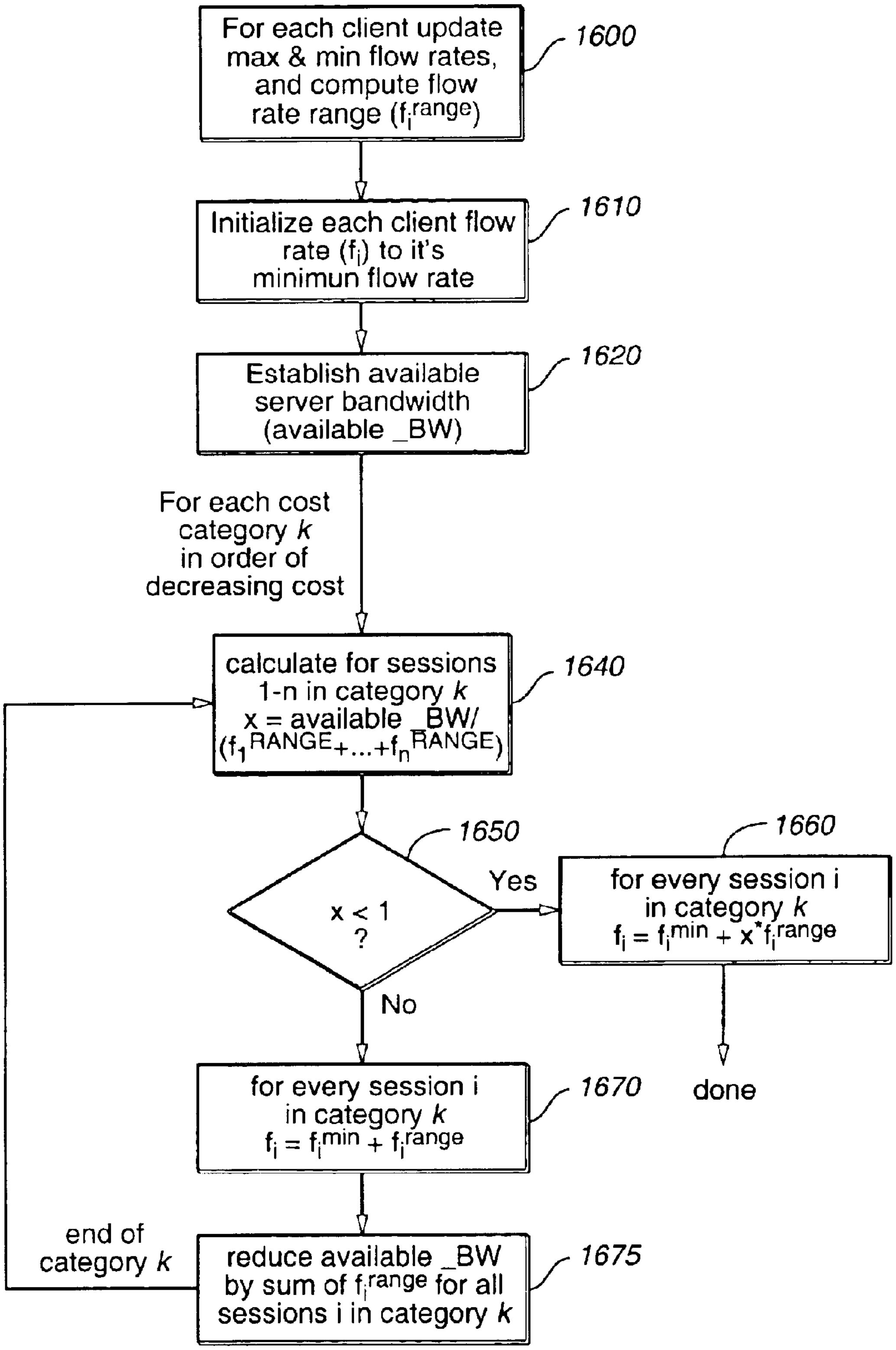

For each client update max & min flow rates, and compute flow rate range (f_i^{range})
1600
Initialize each client flow rate (f_i) to it's minimun flow rate
1610
Establish available server bandwidth (available _BW)
1620
For each cost category *k* in order of decreasing cost
calculate for sessions 1-n in category *k* x = available _BW/ ($f_1^{RANGE}+...+f_n^{RANGE}$)
1640
x < 1 ?
1650
Yes
No
for every session i in category *k* $f_i = f_i^{min} + x^* f_i^{range}$
1660
done
for every session i in category *k* $f_i = f_i^{min} + f_i^{range}$
1670
end of category *k*
reduce available _BW by sum of f_i^{range} for all sessions i in category *k*
1675

FIG._16

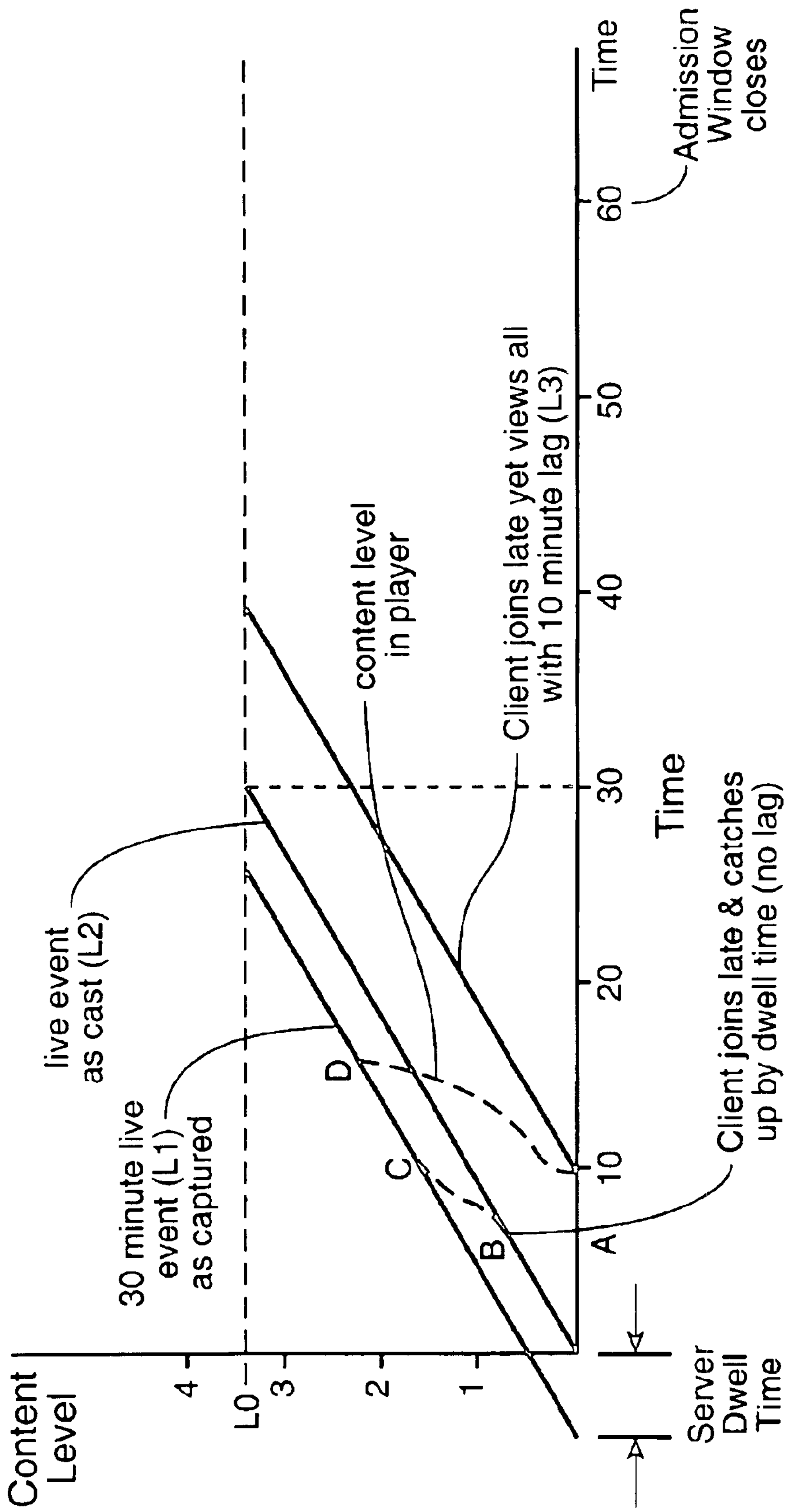

Content Level
4
L0
3
2
1
Server Dwell Time
A
B
C
D
10
20
30
40
50
60
Time
Time
Admission Window closes
30 minute live event (L1) as captured
live event as cast (L2)
content level in player
Client joins late yet views all with 10 minute lag (L3)
Client joins late & catches up by dwell time (no lag)

FIG._17

METHOD FOR CONNECTION ACCEPTANCE AND RAPID DETERMINATION OF OPTIMAL MULTI-MEDIA CONTENT DELIVERY OVER NETWORK

RELATED APPLICATIONS

The present application is a continuation-in-part of U.S. patent application Ser. No. 09/344,688 filed on Jun. 25, 1999 which claimed priority to U.S. Provisional Patent Application No. 60/108,777 filed on Nov. 17, 1998.

BACKGROUND OF THE INVENTION

1. Field of the Invention

This invention relates to the field of delivery of multimedia content over a variety of networks. More specifically, it pertains to multimedia servers which service many clients simultaneously for the delivery of multimedia content which is used and played back at each client. It addresses methods for determining optimal delivery rates to each client and methods for determining whether new clients may be accepted without diminishing the quality of service to existing clients.

2. Status of the Prior Art

In the history of multimedia program delivery, some in the industry have long advocated the use of large client-side buffers and faster-than-real-time content delivery over a network as offering the best of all worlds: a jitter-free viewing experience and a cost-effective utilization of the network resources at hand. Few systems, however, go very far in addressing how to schedule clients or a method for accepting new clients. Real-time systems, often known as streaming systems, can schedule new clients in a very simple manner—if sufficient bandwidth remains for the added real-time stream, then the client may be accepted. However, such systems do not maximize the number of simultaneous clients. On the other hand, faster than real-time delivery, sometimes known as store-and-forward systems, opens up the possibility for more flexible scheduling procedures to control and optimize the number of simultaneous clients while ensuring a high level of quality of service.

The methods for such call acceptance and flow modulation that have been proposed in the prior art have been largely ad-hoc and also incomplete. These have been ad-hoc in the sense that there has been no guiding rationale for their selection from among many possible and potentially superior alternatives. The methods have also been incomplete insofar as they did not address the question of whether any given incoming request for service should be accepted or denied. Video-on-demand systems, or more generally, any system in which a multimedia server is designed to serve multiple clients over a network to deliver bounded content, can benefit from the use of such flow modulation techniques and call acceptance procedures.

Optimal Content Flow Modulation

One time-honored way of designing methods of the class required here is to re-cast the problem to be solved as an optimization problem, in which one seeks to maximize a designated value function moment-by-moment, subject to a set of real-world operational constraints which will typically vary over time. Accordingly, given a set of clients and associated sessions, an optimal delivery procedure continuously establishes content flow rates from the content server to each of its clients so as to maximize aggregate value according to the governing value function.

This approach holds several advantages: 1) optimization problems are well understood, and are tractable by a large and diverse collection of computational methods; 2) if it exists, a global solution that is obtained is arguably optimal by construction, and thus superior or equal to all other.

The present invention teaches the method of optimizing two particular value functions:

1) total data delivered (maximize throughput).

2) total delivery charges (maximize charges).

The first value function does not distinguish one customer from another and will deliver as much data as possible from server to clients irrespective of the characteristics of the latter. The second value function favors the service of high paying customers. It can easily be seen that the first function is a special case of the second one whereby all clients are charged equally.

As will be seen in this disclosure, optimizing for these functions and identifying the necessary constraints requires a new and unique perspective that is specifically designed for the multimedia environment. Moreover, the disclosed methods are specifically designed to account for and accommodate real-world scenarios of today's networks. Consequently many variations of the method are presented to accommodate various scenarios.

SUMMARY OF THE INVENTION

Call/Connection Acceptance Control (CAC)

A CAC procedure is responsible for deciding whether a candidate for service can be accommodated without jeopardizing sessions already in progress at the present time or at some time in the future; failing that it must decide whether a service request should be queued for a time or rejected.

Flow Modulation

Flow modulation methods are those portions of the system which manage the communication and data flow between the server and the clients. Collectively, these methods provide the multimedia data to the client and provide the server with the information about the state of the transmission, playback, user status and network status. These parameters are further used by the present invention in the CAC procedures. In fact, as will be shown, the proposed CAC procedures are tightly integrated with the flow modulation methods.

Adaptation to Variations in Network Capacity

Operational constraints may change over time. For instance, one might elect to vary the total bandwidth available for multimedia content delivery according to the time of day. Alternatively, exogenous data flows on the network may cause unexpected disturbances by usurping available bandwidth and impeding the delivery of data along established session channels. The content delivery strategy of the present invention includes the ability to adapt to scheduled as well as unexpected disturbances so as to minimize unwanted disruptions of services.

Burst Transmissions Provide the Opportunity to Adapt

The present invention, due to it's faster-than-realtime transmissions (also know as burst transmissions), which are realized by use of high-bandwidth networks and large client cache or intermediate storage, provides an opportunity to adapt to changing network conditions. In contrast real-time (streaming) systems are essentially designed for worst-case scenarios: each client must be assumed to constantly use the complete real-time playback bandwidth. Such a system is unable to adapt to any derivation from this scenario. For example, take the simple case where the total server bandwidth is 100% utilized by all clients playing back the streaming video. Should any network condition change, such as a temporary decrease in available bandwidth over the network, then one or more clients' playback is

interrupted, and the system can not recover from such a condition until the bandwidth is regained. Even worse if a single client presses pause either that unused bandwidth must remain reserved and no more clients can be accepted, or that paused client is pushed out in order to service the new client. In essence little or no CAC procedure may be implemented.

In contrast the present invention burst transmits portions of a program and immediately 'gets ahead of itself', thus allowing headroom for a myriad of methods to intelligently handle new clients, client interactivity and possible network fluctuations.

Methods are taught for optimally determining the flow rate to each client. Methods are also taught for accepting or rejecting new clients; these call-acceptance methods are tightly coupled with said flow rate modulation methods. A series of constraint expressions are presented which govern the methods for determining the flow rates and acceptance of new clients. Linear programming techniques are used to optimally solve these expressions. Various embodiments are presented including scenarios for multiple-rate tariffs, and time-of-day bandwidth variations.

BRIEF DESCRIPTION OF THE DRAWINGS

These as well as other features of the present invention will become more apparent upon reference to the drawings wherein:

FIG. **1** depicts the flow of control and/or data between the different stations of a content delivery session;

FIG. **2** illustrates the Entity Data Model;

FIG. **3** geometrically illustrates the linear programming problem statement;

FIG. **4***a* geometrically illustrates an expansion of the flow optimization problem statement;

FIG. **4***b* geometrically illustrates the method for rapid determination of flow rates to maximize flow;

FIG. **5** illustrates a method for implementing flow modulation to maximize flow;

FIG. **6** illustrates a method for implementing flow modulation for maximized charges;

FIG. **7** illustrates typical content flow;

FIG. **8** illustrates typical server swing capacity;

FIG. **9** illustrates a method for call-acceptance and control (CAC);

FIG. **10** illustrates planned constraints on maximum flow;

FIG. **11** illustrates a method for call-acceptance and control (CAC) with scheduled flow changes;

FIG. **12** illustrates stratification of services;

FIG. **13** illustrates a method for call-acceptance and control (CAC) for maximal charge;

FIG. **14** is a system block diagram for a system implementing the present optimization of client bandwidth;

FIG. **15** is a flowchart showing an algebraic method for calculating the optimization of client bandwidth;

FIG. **16** is a flowchart showing an algebraic method for calculating the optimization of client costs; and

FIG. **17** illustrates optimized viewing of a live event.

DETAILED DESCRIPTION OF THE INVENTION

Data & Control Flows (FIG. **1**)

FIG. **1** depicts the flow of control and/or data between the different stations of a content delivery session. As shown a client attempts a connection **100** and manifests itself to the Content Selection subsystem by means of a low bandwidth control channel (not shown). Next the client is authenticated and a selection is made **110**, typically with the aid of browser software. If the client is not authenticated, it is dismissed from the system **120**. If the client has been authenticated and a program selected for viewing then the rate of service is set at this point **130**, perhaps according to the selection that was made, or some contractual stipulation. The client is now placed on the service queue of the CAC subsystem **140**. A client that is made to wait too long will eventually balk **150**. Assuming this does not occur, the CAC subsystem **140** will eventually allocate a channel to the client and open a session **160**. Control now devolves upon the Content Flow Modulator (not shown) which starts the flow of content from server to client **170**. Subsequent capacity changes, whether predictable or not, may force an abrupt termination of a session in progress **180**. Otherwise the session runs to completion **190**.

Entity Data Model (FIG. **2**, listing, table)

The entities entering into our discussion are depicted in FIG. **2**. Client **200** maintains certain data associated with this entity; as shown but not labeled, which includes without limitation, status, id and costOfService. The other entities also each include unlabeled but depicted data. The diagram further depicts the relationship between each entity. As shown, client **200** is assigned a session **240**. Client **200** employs a channel **210**. Client **200** selects contentSelection **230**. Session **240** delivers content through channel **210**. Server **220** modulates channel **210**. Server **200** contains contentSelection **210**. Server **220** accepts, defers or denies client **200** and contentSelection **230** is associated with session **240**.

Furthermore FIG. **2** depicts the various one-to-many relationships. Each client **200** employs one channel **210**. Client **200** may or may not receive one of channel **210**, as depicted by the 0/1 notation. Similarly, client **200** may or may not receive a session **240**. However, whenever client **200** does receive a session **240**, it will always receive a channel **210** since channel **210** and session **240** are allocated as a pair. One or more (N) of client **200** may select one of contentSelection **230**. And server **220** contains one or more (N) of contentSelection **230**. Each one of contentSelection **230** is associated with 0-N of session **240**. Each session **240** delivers content through one of channel **210**. And server **220** modulates one or more (N) of channel **210**.

A more detailed list of each entity of FIG. **2**, and each one's associated description, data elements and function calls is listed below. This listing closely resembles that of object-oriented programming. As such, 'methods' represent the ability to obtain or modify data, while 'attributes' represent data which is directly associated with that particular entity. The listing also includes information relating to one embodiment wherein software programming specifics are disclosed, such as a variable type (double, int and so forth) and more. The present invention is not limited to such an embodiment and other implementations are possible without deviating from the scope and intent of the present invention. The listing, however detailed, is merely illustrative of the data and functions which are used in the equations and methods described herein.

Consequently, data and functions from this listing, associated with the various entities, will be used in forthcoming equations, flowcharts and methods. The reader is directed to this listing as reference when reading such equations and examining such drawings.

—start of entity data model detailed listing—

Model: Untitled 1 (public)

Contains:

client, session, channel, server, contentSelection.

Component: Client (Public Class/Interface)

Comment:

A client entity stands for a client presently requesting or receiving service.

Methods:

public static lookup (id: in int): client

public GetId ( ): const int&

public SetId (val: in int&)

public GetCostOfService ( ): const double&

public SetCostOfService (val: in double&)

Attributes:

private status: client <int>

Specifies whether or not a client has been allocated a channel and session.

private id: int

Integer-valued identifier that is unique to the client (primary key). Can be obtained from a monotonically increasing counter.

private costOfService: double

Dollar charge per Mbyte. This value is the same for all customers under flow optimization. Under cost/charge optimization may be an integer value reflective of the rank; the higher the rank the higher the charge.

Has:

public selected: contentSelection

public assigned a: session

public employs: channel

Component: Session (Public Class/Interface)

Comment:

A session entity holds various state information about the service being received by an associated customer.

public GetCurrentPosition ( ): const double&

public SetCurrentPosition (val: in double&)

public GetPayloadToGo ( ): const double&

public SetPayloadToGo (val: in double&)

public GetStatus ( ): const int&

public SetStatus (val: in int&)

public GetMinimumFlowRate ( ): const double&

public SetMininumFlowRate (val: in double&)

public GetFlowRateRange ( ): const double&

public SetFlowRateRange (val: in double&)

public GetMaxFlowRate ( ): const double&

public SetMaxFlowRate (val: in double&)

Attributes:

private playTimeToGo: double

Indicates the minutes remaining in the viewing experience. Initialized to contentSelectiont.playTime (see below).

private currentPosition: double

Pointer into media content from which content is being delivered.

private payloadToGo: double

The amount of media content (in Mbytes) as yet undelivered by the server. Does not include any content presently stored in the client-side buffer.

private status: int

Indicates whether session is active or paused.

private minimumFlowRate: double

This is the minimum flow from server to client required to ensure uninterrupted service over the remaining playTime. Has a value of zero if playloadToGo is zero. Given by (payloadToGo*8)/(playTimeToGo*60)

private flowRateRange: double

Specifies the effective range over which the channel content flow serving a session is constrained without consideration for interactions with other flows.

Equals maxFlowRate−minimumFlowRate

private maxFlowRate: double

Effective maximum bound on flow as expressed in formula (8) which must be re-evaluated periodically.

Has:

public delivers content through: channel

Component: Channel (Public Class/Interface)

Comment:

A channel represents the network resources from server to client associated with an ongoing session, encompassing the client-side buffer if any, and its level.

public GetBufferLevel ( ): const double&

public SetBufferLevel (val: in double&)

public GetFlowRate ( ): const double&

public SetFlowRate (val: in double&)

public GetMaxFlowRate ( ): const double&

public SetMaxFlowRate (val: in double&)

Attributes:

private bufferSize: double

Capacity of the client-side buffer (or equivalent)

private bufferLevel: double

Current buffer level in MBytes of stored content.

private flowRate: double

Flow rate through channel specified by the relevant optimizing flow modulator.

private maxFlowRate: double

This value represents the maximum possible flow rate from the server to an individual client over its "channel". This value reflects restrictions on flow that pertain to an individual client. It may be determined by factors such as the bandwidth of client's link to the network, or a limit imposed administratively to ensure balanced network utilization.

Component: Server (Public Class/Interface)

Comment:

Entity representing the media server and its CAC and flow modulation activities.

public GetFlowRate ( ): const double&

public SetFlowRate (val: in double&)

public GetMaxMinFlowRate[ ] ( ): const double&

public SetMaxMinFlowRate[ ] (val: in double&)

Attributes:

private maxFlowRate: double

Maximum possible content flow that is allocated to the server by the network.

private flowRate: double

Aggregate content flow rate, summed over all sessions and their associated channels.

private cac__SafetyMargin: double

Tunable safety margin used by the CAC algorithm to protect sessions-in-progress from being affected by changes in available network bandwidth.

private maxMinFlowRate[ ]:double

Applies when N rate tariffs exist. This array holds the maximum floor level for each category of service. The value for the costliest category N is stored in maxMinFlowRate[N−1], and for the least costliest in maxMinFlowRate[0].

It is the relative magnitude of these ascending values that matters, not their absolute value. Thus the actual maximum floor flow rate for category k is given by server.maxFlowRate*(server.maxMinFlowRate[k−1]/server.maxMinFlowRate[N−1]). Similarly, the maximum floor flow rate for category N is server.maxFlowRate.

Has:

public contains: contentSelection

public modulates: channel

Component: ContentSelection (Public Class/Interface)

Comment:

Entity represents a video/sound clip or other bounded unit of content. A continuous data feed does not qualify.

Attributes:

private averagePlayRate: double

The average rate at which media content is consumed by the client, as computed by dividing the (payload*8) by the (playTime*60)

private playTime: double

Duration of play of the media content in minutes.

private payLoad: double

total size of the content in Mbytes.

Has:

public is associated with: session

—end of entity data model detailed listing—

The following table summarizes the highlights of the previous detailed description of each entity in FIG. **2**.

TABLE 1

Entity	Description
client 200	Each client is denoted by an associated unique integer index_{Id}. The set of active clients is denoted by $S_{ActiveClients}$. The set of deferred clients is denoted by $S_{QdClients}$. Incoming clients are expected to select the content they wish to view prior to being queued for dispatch by the CAC sub-system, which requires knowledge of the client's bandwidth requirements, duration of play, and cost of service, all of which may vary according to the selection.
server 220	Servers sit astride a network and can deliver media content through the network to their clients up to a designated maximum flow rate. The server is responsible for accepting or rejecting clients, launching sessions and associated channels for the former, and modulating content flows over all channels in an optimal manner.
channel 210	A channel represents the data path between the server and the client. The channel buffer is typically located near or within the clients viewing station. The flow of content through the channel is set by the flow modulator sub-system.
contentSelection 230	A server will typically act as a repository for media content, which it can deliver to clients upon demand. For our purposes, media content is characterized by its payload and the play duration, which together imply the averagePlayRate = (payload*8)/(playTime *60). The averagePlayRate is none other than the streaming rate imposed by real-time just-in-time streaming algorithms.
session 240	Every session represents an instance of media content delivery to an associated client over a designated channel. The playTimeToGo indicates the time remaining before the content is fully played out to the client. The payloadToGo is the amount of content data as yet undelivered to the channel. A session terminates when this value reaches zero, at which time playTimeToGo may still be large, according to the capacity, the level of the channel buffer, and the media play rate.

Constraints on Content Flow

Before referring to more Figures, it is imperative to establish some formulas and problem statements which are used in the methods which follow.

The flow of content between entities is subject to the following constraints at all times. Buffer levels are always expressed in Mbytes and data rates in Mbits/sec.

$$\Sigma_{i \in S_{activeClients}}(\text{client.lookup}(i)\text{.channel.flowRate}) <= \text{server.maxFlowRate} \quad (1)$$

The sum of all channel flows cannot exceed the imposed maximum throughput capacity of the server.

$$\text{client.lookup}(i)\text{.channel.flowRate} <= \text{client.lookup}(i)\text{.channel.maxFlowRate for all } i \in \Sigma_{activeClients} \quad (2)$$

The data path from server to client is subject to its own constriction.

$$\text{client.lookup}(i)\text{.channel.flowRate} <= (\text{client.lookup}(i)\text{.channel.bufferSize} - \text{client.lookup}(i)\text{.channel.bufferLevel})*8/60 + \text{client.lookup}(i)\text{.session.mediaContent.averagePlayRate, for all } i \in \Sigma_{activeClients}, \quad (3)$$

The channel buffer is never allowed to overflow.

$$\text{client.lookup}(i)\text{.channel.flowRate}<=\text{client.lookup}(i)\text{.session.payloadToGo*8/60 for all } i\in\Sigma_{activeClients}, \quad (4)$$

Content that does not exist cannot be delivered. (Constraint 1 will ordinarily prevail except at the very end of a session.)

The constraints listed above are straightforward applications of common sense in relation to the flow of data through constricted channels, out of finite data sources, and into and out of bounded buffers. By contrast, the following constraint, which imposes a minimum channel flow rate instead of a maximum, is less obvious. The minimum value, termed the minFlowRate is set to the flow rate which, if sustained over the balance of the play time to go (playTimeToGo), ensures that all required content will be available when needed—and no sooner—until all content is played out. This floor value can be calculated for $i\in\Sigma_{activeClients}$ by the formula:

$$\text{client.lookup}(i)\text{.session minFlowRate=(client.lookup}(i)\text{.session.payloadToGo*8)/(client.lookup}(i)\text{.session.playTimeToGo*60)} \quad (5)$$

Accordingly, the flow rate on the right-hand-side is termed the just-in-time (JIT) flow rate (f^{JIT}).

Thus:

$$\text{client.lookup}(i)\text{.channel.flowRate>=client.lookup}(i)\text{.session.minFlowRate for all } i\in\Sigma_{activeClients} \quad (6)$$

The variable constraint bounds (i.e. the values to the right of the inequality symbol) of equations 1–4 and 6 are re-evaluated on a periodic basis (e.g. once per second) prior to the execution of the CAC procedure and optimizer. In particular, the minFlowRate value starts out at the beginning of a session equal to the streaming rate. By construction the minFlowRate rate never exceeds this initial value so long as constraint 6 is honored. In fact, constraint 5 implies that the minflowRate value must be a diminishing function of time that may hold its value for a time but never rises. As seen from equation 6, the actual data rate of the channel, flowRate, is always greater than or equal to the minFlowRate. By design, and virtue of the fact the present invention uses faster-than-realtime transmissions, the system quickly gets ahead of itself and ensures that after initial conditions, the minFlowRate is always equal to or less than the real-time rate and that it continues to decrease. As we shall see, the CAC procedure exploits this monotonic characteristic of the minimum flow rate over time.

Constraints 2, 3 and 4 are of like kind, each specifying an upper bound on individual channel flows. Whereas the bound for constraint 2 is typically a constant, the bounds on 3 and 4 will vary over time. Nevertheless, only one of the three bounds is effective at any given time, namely the one with the smallest bound value, given by:

$$\text{client.lookup}(i)\text{.session.maxFlowRate=minimum of} \quad (7)$$

1) client.lookup(i).channel.maxFlowRate,
2) (client.lookup(i).channel.bufferSize−client.lookup(i).channel.bufferLevel)*8/60+client.lookup(i).session.mediaContent.averagePlayRate,
3) client.lookup(i).session.payloadToGo*8/60

Consequently, formulas 2, 3, and 4 can be consolidated into a single constraint, the bound for which is computed at every scan to be the smallest bound of associated constraints 2, 3 and 4.

$$\text{client.lookup}(i)\text{.channel.flowRate}<=\text{client.lookup}(i)\text{.session.maxFlowRate}, \quad (8)$$

whereby for all $i\in\Sigma_{activeClients}$, maxflowRate is given by equation (7).

At any one time, individual channel flows are constrained over a range, as follows:

$$\text{client.lookup}(i)\text{.session.flowRateRange=client.lookup}(i)\text{.session.maxFlowRate}-\text{client.lookup}(i)\text{.session.minimumFlowRate} \quad (9)$$

Value Functions

The value functions introduced previously can be expressed mathematically as linear functions of channel flows, as follows:

Optimizing Throughput (Maximal Flow)

$$\text{value}=\Sigma_{i\in SactiveClients}\text{Client.lookup}(i)\text{.channel.flowRate)} \quad (10$$

Optimizing Charges (Maximal Charges)

$$\text{value}=\Sigma_{i\in SactiveClients}\text{(client.lookup}(i)\text{.channel.flowRate*client.lookup(i).costOfService)} \quad (11)$$

Optimization Problem Statement (FIG. **3**)

The optimization problem, which in one embodiment is strictly throughput and in another case is charge, can be stated simply as follows:

Find values for

client.lookup(i).channel.flowRate for all $i\in\Sigma_{activeClients}$

constrained by inequalities 1 through 5, such that the value obtained by evaluating expression 10 or 11 assumes a maximum.

Both of these problem formulations are examples of Linear Programming for which a number of well-known and generally effective computational solutions exist. In linear programming one seeks to optimize a linear cost function of variable x

$$c*x=c_1*x_1+\ldots+c_n*x_n \quad (12)$$

subject to a set of linear inequality constraints

$$A*x<=b \quad (13)$$

where $x^T=(x_1,\ldots,x_n)$, $c=(c_1,\ldots,c_n)$ are the state variable & cost vectors, A is an n-by-m matrix, $b^T=(b_i,\ldots b_m)$ is the constraint vector, and the operator '*' stands for matrix or scalar multiplication.

FIG. **3** is introduced as illustrative of the problem statement and the general methods of the prior art, and is not incorporated as an element of the invention.

The linear programming problem as well as its solution can best be understood with the aid of geometry. FIG. **3**, depicting a 2-dimensional Cartesian problem space, inequality constraints (13) define a convex hull H **310** over which a search for an optimum value of $x=(x_1,x_2)$ is permitted to range. The cost vector c **350** defines an infinite family of equal cost lines (hyper-planes) which lie orthogonal to c. Three examples of such lines are shown in L_1 **360**, L_2**365**, and L_3 **370**., each of progressively higher value. The supreme value of the cost function is obtained by sliding along c **350** till one can go no further, in this instance toward vertex V_4 **340** of hull H **310**. Many well-known methods (e.g. the Simplex Method) work roughly in this fashion, exploiting the fact that at least one optimum point must be at a vertex. In particular, the Simplex method algorithm

begins by finding a vertex (e.g. V_2 **320**), and then moves along a sequence of vertices (e.g. V_3 **330**, V_4 **340**) improving itself each time until no further improvement is possible & the summit is reached.

Let us suppose instead that V_3 **330** were placed along L_3 **370** along with V_4 **340**. According to prior art methods, V_3 **330** and V_4 **340** are the two possible solutions, but the equally valuable points in between them are not. As we shall soon see, the problem of throughput optimization (6) falls in this category.

While vertex V_1 **300** does not factor into this description, it is depicted in FIG. **3** for completeness.

Flow Modulation

Methods for Maximal Flow

The following sections detail two embodiments to optimize total data flow.

Overview (FIG. **4**-*a*)

FIG. **4**-*a* depicts a scenario involving two flows. The convex hull is in this instance bounded by line segments L**1**, L**2**, L**3**, L**4** and L**5**. L**6** is a boundary used in a different embodiment, however the present embodiment uses L**5** and not L**6**. Flow f_2 can range over the interval separating line segments L**1** from L**3**, namely f_2^{MIN} and f_2^{MAX}; the range is depicted as f_2^{RANGE}. Flow f_1 can range over the interval between lines L**2** and L**4**, namely f_1^{MIN} and f_1^{MAX}, and depicted as f_1^{RANGE}. The sum of flows f_1 and f_2 is constrained to lie inside of line segment L**5** which, by construction, is always orthogonal to the cost vector C_f. Cost vector C_c is also illustrated but used in a distinct embodiment. In the present embodiment only C_f is used. In the illustrated example of the present embodiment, the constraint on total flow is set to 5, and is therefore low enough to cause L**5** to intersect L**3** and L**4**. This would not have been true had the value chosen had been 10 instead of 5. With L**5** safely out of contention, the convex hull would instead be a simple rectangle bounded by L**1** through L**4**, thereby permitting both flows to assume their respective maxima without interference. In practice operational constraints exist intrinsically or are imposed from the outside so as to ensure cost effective sharing of potentially costly network resources.

Supposing FIG. **4** to be correct, the well-known methods would select vertex $V_{3\text{-}5}$, which lies at the intersection of L**3** and L**5**, or $V_{4\text{-}5}$, which lies at the intersection of L**4** and L**5**. These solutions, though optimal, are undesirable for the present invention as they fail to spread available bandwidth over all channels as fairly as would a centrally located interior point of L**5**. For this reason, two alternative optimization methods are taught, which are adapted to the particular needs of this problem and ensure a fairer allocation of constrained bandwidth among all channels.

Iterative Procedure (FIG. **5**)

In order to optimize use of all available bandwidth, the following general method is used, with the details illustrated in FIG. **5**. This method is a solution for the problem illustrated in FIG. **4**-*a*, which geometrically illustrates the optimization problem in the limited case of two flows, f**1** and f**2**. The following description expands the problem to an arbitrary number of clients (and therefore flows) and presents a method for solving this optimization problem.

Referring to FIG. **5**, in step **500**, values are calculated for the session maxFlowRate and session.minFlowRate for each client as per the minimum and maximum constraint bound expressions in equations 6 and 8, respectively. This step correlates to the determination of $f_1$min, $f_1$max, $f_2$min and $f_2$max from FIG. **4**-*a*.

The difference between these two yields the session.flowRateRange of each client. Thus:

session.flowRateRange=session maxFlowRate−session.minimumFlowRate

In step **505**, the active clients are sorted in an ascending fashion based upon their session.flowRateRange. As will be shown this critical step facilitates allocation of the remaining server bandwidth as evenly as possible among all active channels, thus maximizing the number of channels that benefit by use of the total server bandwidth. An arbitrary assignment of remaining bandwidth is likely to saturate the server before all channels have been assigned extra bandwidth, thereby favoring certain channels on an ad-hoc basis. This step correlates to keeping the solution in bounds of the space delinated by line segments L**1**–L**5** of FIG. **4**-*a*.

In step **510**, each client's channel flow rate is set to the session mimimumFlowRate.

By doing so, it is ensured that the minimum flow constraint is met for each session and that the minimum flow rate is a non-increasing function of time, which is critical to the proper functioning of the CAC procedure. This portion of the process also ensures that the solution, starting from the vertex in FIG. **4**-*a* as defined by the intersection of L**1** and L**2**, moves generally in the direction of vector c_f. All clients are also marked as unprocessed.

In the next step, **520**, server.flowRate is set to the sum of each active client's session.flowRate.

Next, the following is iterated over all clients in sorted sequence (during any given iteration the selected client is given by its id) by performing steps **530** through **570**. In step **530** evaluating the following expressions test for possible server saturation:

delta=(server.maxFlowRate−server.flowRate)/(qty of un-processed clients)

range=client.lookup(*id*).session.maxFlowRate−client.lookup(*id*).session.flowRate

If range is greater then delta, this implies that the server can be saturated in this iteration by allocating delta to all unprocessed clients (step **540**).

On the other hand, the 'no' path for step **530** implies that the server is not saturated and that the present client (given by id) will saturate first. Accordingly, in **550** the delta variable is set as follows:

delta=range

To again correlate this process back to the geometry of FIG. **4**-*a*, server saturation is indicated when the solution which is being sought in the direction of vector c_f goes beyond line segment L**5**.

Next, the flow rate is incremented for all unprocessed clients by delta, causing client id to saturate.

In step **560** the server flow rate is adjusted accordingly:

server.flowRate=server.flowRate+delta*(qty of unprocessed clients)

In step **570** the client given by id, now saturated, is marked as processed.

Algebraic Procedure (FIGS. **4**-*b* and **15**) By Interpolation

Referring to FIG. **4**-*b*, the iterative method described in the previous section begins its search for a maximum from vertex V_{min} along the direction shown by line C_1. In contrast, the present method follows the diagonal line segment L_{diag} of the rectangle bounded by lines L**1** through L**4**, starting at vertex V_{min} and ending at point $V_{3\text{-}4}$.

Provided it exists, the intersection between L_{diag} and **L5**, indicated by V_f, is optimal on the same basis as any other point lying along the intersection of **L5** with the rectangle bounded by **L1** through **L4**, as previously discussed.

Whenever **L5** does not intersect this rectangle, the optimal solution is given by vertex **V3-4** at which **f1** and **f2** both assume their respective maxima.

In the first instance, the coordinates for point V_f can be obtained by elementary vector geometry and algebraic manipulation, as follows:

First, we define a number of abbreviations,

f_k^{min} represents client.lookup(k).session.minimum flow rate

f_k^{range} represents client.lookup(k).session.flow rate range

f_{svf}^{max} represents server.maxflowrate

We seek scale factor x such that the vector sum of (f_1^{min} . . . f_n^{min})+x(f_1^{range} . . . f_n^{range})intersects an equi-cost hyper-place for capacity c.

The point of intersection is obtained by solving equation:

$$(f_1^{min}+ \ldots +f_n^{min})+x(f_1^{range}+ \ldots f_n^{range})=c$$

Solving for x, we obtain:

$$x=[f_{svr}^{max}-(f_1^{min}+ \ldots +f_n^{min})]/(f_1^{range}+ \ldots +f_n^{range})$$

where $f_{svr}^{max}-(f_1^{min}+ \ldots +f_n^{min})$ represents the unused bandwidth beyond the minimum allocation for each session. For any given session, the optimal flow rate is given by:

$$f_i=f_i^{min}+xf_i^{range}, \text{ if } x<1,$$

or

$$f_i=f_i^{min}+f_i^{range}, \text{ if } x>1$$

This solution by interpolation is interesting by virtue of the great efficiency of the calculation, and the fact that no session saturates (x>=1) unless all sessions saturate. In contrast, the previous method entails considerably more computational effort, and tends to saturate sessions with the lowest reserve capacity.

A block diagram for this algorithm is depicted in FIG. **15**. Block **1500** establishes the maxim and minima for each flow. For each flow the minimum is subtracted from the maximum to obtain the range. Available Bandwidth is calculated in block **1520**, as the difference between the aggregate flow capacity and the sum of flow minima. Factor x is calculated next in block **1540**, and tested against 1 in block **1650**. Flow rates for each session are computed in block **1560** if x is less than 1, or **1570** in the contrary case.

A Method for Maximal Charge

The following sections detail one embodiment to optimize the total monetary charges within the system. The second method is algebraic in nature, and thus very efficient.

Overview (FIG. **4**-*a*)

Referring back to FIG. **4**-*a*, cost vector C_c lies orthogonal to line **L6**, which intersects the convex hull at the vertex formed by the intersection of lines **L4** and **L5**, namely $V_{4\text{-}5}$. This cost vector, and the optimal point that it implies, favors flow **f1** over flow **f2**. In this example, as the cost of service for **f1** equals 2, thus exceeding the cost of service of 1 set for **f2**. As the number of flows grow to exceed the number of distinct categories of service (and associated costs of service) the unique optimal solution, depicted in FIG. **4** for the case where every flow has a distinct cost of service, no longer applies. Once again a plurality of flows within a service category vie for bandwidth which a method should endeavor to distribute evenly. This method is derived from

the previous one, and optimizes one cost category after another, starting with the most costly and ending with the least costly, or when all available capacity is allocated.

An Iterative Search Procedure (FIG. **6**)

Let the service categories be denoted by k=1 . . . N, where k also denotes the cost of service.

Let $C_1 \ldots C_N$ be the partition of $S_{activeClients}$ that places all clients with cost of service k in set C_k. Partition sets C_k can be ordered to form sequence SeqC=$C_N \ldots C_1$.

FIG. **6** depicts the method for implementing the method to maximize the cost of service (service charge).

This method is nearly identical to the iterative procedure used to maximize flow. The principle difference stems from the partitioning of clients according to their category (cost) of service: clients charged most are allocated bandwidth preferentially. This is accomplished by adding another level of iteration around the method of FIG. **5**. The inner iteration (steps **650** through **680**) functions exactly as before, with the difference that its actions are limited to the clients belonging to the given service category k (i.e. C_k). This difference also holds true of step **640** which sorts category k clients according to their flow ranges prior to entry in the bandwidth-allocating inner loop. The outer loop proceeds down a sorted sequence of service categories SeqC (generated in step **630**), starting with the category generating the greatest revenue to the service provider. Given a fairly static set of service categories, this sort need be performed only when the categories undergo change. Steps **670**, **675** and **680** are identical to their counterparts in the method of FIG. **5** (i.e. **570**, **575** and **580**).

The net effect of this method is preferential allocation of bandwidth according to category of service, and equitable treatment of clients within the same category of service.

An Iterative Algebraic Procedure (FIG. **16**)

The algebraic method used to maximize bandwidth (FIGS. **4***b* and **15**) can be viewed as a special case of cost optimization in which all costs are equal. By construction, all sessions belonging to a given category, starting with the most expensive category, and ending with the least, and updating available bandwidth after each step in sequence, we obtain an iterative algebraic method that scales much better than the preceding method.

A block diagram for this algorithm is depicted in FIG. **16**. Block **1600** establishes the maxima and minima for each flow. For each flow, the minimum is subtracted from the maximum to obtain the range. In block **1610**, every session flow is assigned the minimum allowed value, which value will apply by default should the iterative procedure run out of bandwidth before all categories can be considered. Block **1620** computes the initial value for available bandwidth, as the difference between the aggregate flow capacity and the sum of flow minima.

Flow values are obtained for each cost category in order of decreasing cost within blocks **1640** through **1675**. Factor x is calculated next in block **1540**, and tested against 1 in block **1650**. Flow rates for each sessions of the category under consideration are computed in block **1660** if x is less than 1, or **1670** in the contrary case. In the former case there is no more bandwidth to allocated to the sessions in the category under consideration.

Call Acceptance Control (CAC)

CAC for Maximal Flow Overview (FIGS. **7–8**)

The CAC procedure applicable to this flow optimization relies on the essential step of accepting a new client if and only if the added load induced thereby does not compromise service to existing clients or the new one. This critical step could not be accomplished without the close integration with previously-described flow-modulation methods of FIGS. **5**, **6**, **15** and **16**.

According to the previous discussion, the minimum flow rate is the minimum sustained flow rate that guarantees that the associated viewer will not be subject to interruptions in service due to a shortfall of content from the media server. It follows that whenever data is being delivered at a rate in excess of the minimum flow rate, a downward adjustment toward the minimum level could be accommodated as needed to surrender bandwidth to any newcomer.

FIG. **7** depicts content flow over a channel for the course of a typical session, and also how data is delivered under real-time streaming D. The amount of content delivered is the same in either case, but the manner of delivery differs considerably. A session is launched at time 0 as the network is lightly loaded, and the optimizer sets an accordingly high flow rate. Another client emerges at the end of interval **700**, causing a downward adjustment to the flow rate over interval B, as available bandwidth is shared between two sessions. During both of these intervals the minimum flow rate **720** drops quickly, as data accumulates in the client's media buffers. At the end of interval B, a massive influx of clients necessitates that flow be dropped to the minimum flow rate, which now lies substantially below the streaming rate D and is held until all data is delivered at the end of interval C. Note that the minimum flow rate, shown as element **720**, diminishes monotonically over time.

The server swing capacity is defined as the difference between the maximum capacity of the server and the total minimum flow rates for all active clients. Therefore:

$$\text{swingCapacity=server.maxFlowRate}-\Sigma_{i\in S_{activeClients}}\text{(client.lookup}(i)\text{.session.minFlowRate)} \quad (14)$$

Given the monotonic decreasing nature of session minimum flow rates, server swing capacity can readily be seen to be a monotonic increasing function of time over the intervals separating client admissions, at which point it undergoes a drop as a new load is taken on. This all-important characteristic implies the following:

Any client admitted for service based on the present value of swing capacity is guaranteed to have sufficient bandwidth at its disposal over the entire future course of the session.

FIG. **8** depicts the server swing capacity **800** over the course of the sessions illustrated in FIG. **7**. Swing capacity rises quickly over intervals A & B as data is delivered at high flow rates over the network. It holds steady over interval C when all channels flow at their minimum rate then jumps at the end of C before resuming its monotonic rise once again.

Procedure (FIG. **9**)

In this procedure, which executes on a periodic basis, queued clients awaiting bandwidth are scanned in FIFO order. For each one, the required bandwidth is computed as per the client's prior content selection. If the available swing capacity (reduced by a safety margin) exceeds the amount required, then the client is activated and swing capacity is adjusted accordingly. Otherwise, two possible cases are considered: 1) under the FirstFit embodiment, the procedure continues scanning clients to the end of the queue, activating clients whose requirements can be met; 2) under the FIFO embodiment, the procedure ends with the first candidate client whose requirements cannot be met.

In step **900**, available server swing capacity is evaluated according to the formula

$$\text{swingCapacity=server.maxFlowRate}-\Sigma_{i\in S_{activeClients}}\text{(client.get}(i)\text{.session.minimumFlowRate)}$$

The bandwidth requirement for client id in Step **920** is obtained as follows: required_bandwidth=client.lookup(id).contentSelection.averagePlayRate

The predicate evaluated in Step **940** is given by the expression:

(required_bandwidth<=swingcapacity−server.cac_flowSafetyMargin)

In step **950**, client activation entails allocation of a session and a channel, and insertion in the set of active clients eligible for bandwidth allocation by the optimal flow modulator.

In step **960** the swing capacity is diminished by the amount reserved for the activated client:

swingcapacity=swingCapacity−required_bandwidth;

Responding to Variations in Network Capacity (Maximal Flow)

In the CAC procedure for maximal flow, a safety margin was introduced, namely server.cac_flowSafetyMargin, to provide the means for ensuring that the server's swing capacity will never fall below a minimal threshold value.

According to this procedure, the following inequality always holds true:

swingcapacity>=server.cac_flowSafetyMargin (15)

In the previous discussion, a server's swing capacity provided the basis for determining whether or not a prospective client should be allocated bandwidth.

By holding server.cac_flowSafetyMargin in reserve, the CAC algorithm forces delivery of content at faster than real-time rates among accepted clients, even under the heaviest load. The net effect is to apply upward pressure on client side buffer levels, thus promoting continuous accumulation of content and a jitter-free viewing experience once a session is under way.

In another embodiment, server swing capacity can also be interpreted as specifying the maximum amount by which the server.maxFlowRate constraint can be dropped without affecting service, should such an adjustment prove necessary due, for instance, to an influx of exogenous network traffic that diminishes the amount available for multi-media services. Parameter server.cac_flowSafetyMargin can thus be set so as to guarantee a minimum capacity to tighten the constraint on maximum server flow in response to unexpected load changes that affect the server's ability to service its existing clients as well as new ones.

Anticipating Scheduled Variations in Network Capacity (Maximal Flow)

Overview (FIG. **10**)

FIG. **10** depicts how the constraint on maximum flow might be allowed to vary according to the time of day, day of the week, and so forth, in expectation of time-varying traffic flow levels extrapolated from past experience, traffic flow models, etc. Maximum flow rate **1000** can be seen to vary based upon the time of day. In practice, defining the right-hand-side of inequality constraint 1 as a time-dependent expression can impose such time-varying capacities. According to the previous description, the optimizer, which executes on a periodic basis, will automatically seek new flow levels for every active session as the constraint varies. There is, however, no guarantee that an acceptable operating point will be found for all sessions (i.e. one that respects the minimal and maximum constraints on session channel flow). One such example is the case where the server is loaded to the limit and total capacity is curtailed in excess of the aforementioned safety margin. Should such a situation arise the only recourse may well be the termination of a number of established sessions (i.e. load shedding).

The goal is to eliminate service disruptions of this sort by allowing the CAC procedure to look ahead into the future, and accept new clients only if these can be accommodated without any compromise in service in the midst of previously anticipated changes in available network capacity. The following CAC procedure generalizes the previous one: before accepting a client, the test on swing capacity is repeated over a sequence of time segments that cover the proposed viewing period.

Definitions

Let:

$$t\_\text{end}(i)=\text{client.lookup}(i).\text{session.playTimeToGo}+t\_\text{now} \quad (16)$$

Let server.maxFlowRate(t) be server flow capacity as a function of time, as exemplified in FIG. **10**.

Let Seq_T(t_now)=advancing sequence of future times, lead by t_now, when server.maxFlowRate(t) undergoes a step change. For instance, at 9:15 in FIG. **10** this sequence reads as follows: 9:15, 9:30, 11:30, 13:30, 6:30, 7:30.

The server swing capacity at a future time t is computed according to the capacity and worst-case client flows at time t.

$$\text{swingcapacity}(t)=\text{server.maxFlowRate}(t)-\Sigma_{i\_end(i)>t}(\text{client.lookup}(i).\text{session.minFlowRate}) \quad (17)$$

It is noteworthy that the worst-case client flows at time t are express terms of the present minimum flow rates, which cannot increase over time, but might hold steady. Finally, a predicate is defined that tests whether a prospective customer will cause swing capacity to be exceeded at some time t, as follows:

```
(18) boolean client__fits(i ,t) {
        if(client.lookup(i).contentSelection.averagePlayRate < =
            swingCapacity(t) - server.cac__flowSafetyMargin)
        return true;
    else return false;
    }
```

Procedure (FIG. **11**)

This procedure is an adaptation of the first, which has been extended to consider swing capacity at times in the future when capacity undergoes scheduled changes. Before accepting a client, its minimal bandwidth requirement (which by construction of the flow modulator will never increase over time) is checked against projected swing capacity at points in time when total available capacity undergoes scheduled step changes, provided these times fall within the proposed content viewing period. A candidate is activated only if all tests succeed.

Step **1100** builds a sequence of time values (SeqT) at which step capacity changes are scheduled to occur. The first element of this sequence is t_now, representing the present.

Beyond step **1100** the queue of waiting clients is scanned in FIFO order, yielding a candidate designated by id at each iteration.

The bandwidth requirement for client id in Step **1120** is obtained as follows:

$$\text{required\_bandwidth}=\text{client.lookup}(id).\text{contentSelection.averagePlayRate}$$

The worst-case end time for content flow to id is obtained according to the content selected, as follows:

$$t\_\text{end}=\text{t\_now}+\text{client.lookup}(id).\text{selected.playTime}$$

Steps **1130** through **1150** are executed within an iteration for each time point t in SeqT falling between t_now and t_end. This iteration is ended in step **1130** if t exceeds the time window of interest, or in step **1150** if the supply of scheduled capacity changes is exhausted.

For each time value, step **1140** compares required bandwidth to projected swing capacity.

Projected swing capacity at time t is:

$$\text{swingCapacity}(t)=\text{server.maxFlowRate}(t)-\Sigma_{i\_end(i)>t}(\text{client.lookup}(i).\text{session.minimumFlowRate})$$

Note that only active clients whose t_end times occur after t are considered in the sum of minimum flow rates.

The predicate expression used in step **1140** at time t is thus:

$$(\text{required\_bandwidth}<=\text{swingCapacity}(t)-\text{server.cac\_flowSafetyMargin})$$

Step **1160** performs the same actions as step **660** in the previous CAC flowchart

The first CAC process is a special case of the present one, in which the set of step change times to server.maxFlowRate is empty (i.e. server.maxFlowRate is constant), and Seq_T(t_now)==t_now.

In preparing Seq_T(t_now), one need only consider future times that will pass before the longest possible content is played out if started at t_now. In order to sidestep problems associated with rollover (at midnight, year 2000, etc.), time is best expressed as a monotonically increasing value (e.g. seconds since Jan. 1 1990).

CAC for Maximal Charges

Overview (FIG. **13**)

Previously, a method for flow modulation was presented that maximizes charges to clients with active sessions. The CAC embodiment presented previously was not sufficient as it does not consider the cost of service as a basis for connection acceptance. As a result, it may turn away higher paying customers while granting service to lower paying ones, thereby defeating the purpose for this particular embodiment. Therefore, another embodiment is defined which offers the following features:

1. Awaiting clients are serviced in order of their respective service categories, higher paying clients first.

2. Once accepted, a client is guaranteed to receive acceptable service irrespective of its service category.

3. Under heavy load conditions higher paying customers are more likely to be accepted than lower paying ones.

4. Lower paying customers will no be starved for service when higher paying ones enjoy a surplus.

5. Available bandwidth is not thrown away needlessly while clients are being denied service.

The first objective is easily met by dividing the client queue into as many bands as there are service categories, resulting in a banded queue. Bands are ordered within the queue according to their service categories, with the costliest category in front. As prospective clients arrive and make their selection they are placed in their respective queue band according to their service category (which may be set contractually, according to content selection, etc.).

Our second objective is met by employing a procedure patterned after those presented previously & offering the same guarantee. Toward our third and fourth objectives we propose dividing total available bandwidth in as many strata as there are service categories. As depicted in FIG. **12**, two service categories are shown, Premium and Basic, each entailing an associated cost of service. A prospective client is accepted only if there is sufficient swing capacity available

within its given service category. The swing capacity for a given category is given by the smaller of 1) the difference between its maximum floor flow rate (corresponding to the top of the stratum for the service category) and the sum of the minimum rates of all active sessions in its category or below, and 2) available swing capacity overall. Finally, our fifth objective is met by allowing the flow optimizer to function freely subject to its operational constraints. The imposed ceilings on call acceptance by category relate to minimum flow rates, which merely impose a floor on actual flow rates. For example, basic clients might well consume all available bandwidth **300** in the absence of any premium customers, yet could be throttled back toward their floor flow rates (which together cannot exceed **200** in this example) at any time should any premium customer suddenly demand service. In contrast, premium customers could consume the entire **300** bandwidth. As lower paying customers appear these would be admitted to the degree that their quota on minimum flow is not exceeded (i.e. **200**) and the availability of swing capacity on the system.

Procedure (FIG. **13**)

The present procedure requires a number of ancillary definitions, which follow:

Let the service categories be denoted by k=1 . . . N, where k also denotes the cost of service.

Let server.maxMinFlowRate[k−1] be the top of the stratum for service category k. Note that server.maxMinFlowRate [N−1]=server.maxFlowRate.

Let S_k be the set of active client indices with a service category equal to or less than k. Note that **S1** is contained in **S2**, **S2** is contained in **S3**, and so forth, and that $S_N=S_{activeClients}$.

Let swingcapacity(k) denote available swing capacity for service category k. By construction:

$$\text{swingCapacity(k)=minimum of: (server.maxMinFlowRate[k−1]−}\Sigma_{i\in S_k}\text{(client.lookup}(i)\text{.session.minFlowRate)),} \quad (19)$$

$$\text{(server.maxFlowRate−}\Sigma_{i\in S_{activeClients}}\text{(client.lookup}(i)\text{.session.minFlowRate)))}$$

Referring to FIG. **13**, this method is used for CAC when multiple rate tariffs are in effect, and there is a desire to maximize economic returns to the service provider while offering acceptable service to all.

All waiting clients are scanned in FIFO sequence. The actions taken in Steps **1320** and **1360** are identical to those described in connection with earlier CAC flowcharts.

Step **1340** evaluates a predicate expression that tests whether the required bandwidth can be accommodated without exceeding the lesser of 1) swing capacity available to the client's category of service, and 2) total available swing across all categories of service. The latter factor could be determinative if all available bandwidth were allocated to high paying customers, leaving lower paying ones such as the proposed client unable to draw from their unfilled quota.

Let us suppose that candidate client id belongs to rate category k.

We define the swing capacity available in rate category k as:

$$\text{swingcapacity}(k)\text{=least of: (server.maxMinFlowRate}[k-1]-\Sigma_{i\ in\ S_K}\text{(client.lookup}(i)\text{.session.minimumFlowRate))}$$

and

$$\text{(server.maxFlowRate−}\Sigma_{i\in S_{activeClients}}\text{(client.lookup}(i)\text{.session.minimumFlowRate))}$$

The predicate expression invoked by step **1340** can now be written as follows:

$$\text{(required\_bandwidth<=swingCapacity}(k)\text{−server.cac\_flowSafetyMargin)}$$

This algorithm processes queued clients in band sequence, and within every band in FIFO order. If the predicate evaluates to true, the client is activated. Otherwise two possible cases are considered: 1) under the FirstFit embodiment, the procedure continues scanning clients to the end of the banded queue, activating clients whose requirements can be met; 2) under the FIFO embodiment, the procedure ends with the first candidate client whose requirements cannot be met. Many other variations on these two embodiments might also be considered.

Anticipating Scheduled Variations in Network Capacity (Maximal Charge) Overview

The procedure applicable to optimization of delivery charges is obtained by blending elements of the CAC method depicted in FIG. **13** into the method depicted in FIG. **11**, which applies without change. To understand how this might work it may be useful to visualize a version of FIG. **10** stratified along its length in the manner of FIG. **8**. As the maximum flow level undergoes a step change, so too do the widths of its constituent strata in equal proportion.

Procedure

As previously mentioned, the CAC method (FIG. **11**) applies to this circumstance also, provided we alter the definition of two routines, **(17)** and **(18)**, upon which that procedure relies, yielding **(20)** and **(21)**, and adopt the banded queue organization outlined in the previous section.

The server swing capacity at a future time t is computed according to the capacity and worst-case client flows at time t.

$$\text{swingCapacity}(k,\ t)\text{=minimum of((server.maxFlowRate}(t)\text{*(server.maxMinFlowRate}[k-1]\text{/server.maxMinFlowRate}[N-1])\Sigma_{i\ e\ Sk\ \&\ (t\_end(i)>t)}\text{(client.lookup}(i)\text{.session.minFlowRate)),} \quad (20)$$

$$\text{(server.maxFlowRate−}\Sigma_{i\in S_{activeClients}\ \&\ (t\_end(i)>t)}\text{(client.lookup}(i)\text{.session.minFlowRate))}$$

Finally, we define a predicate that tests whether a prospective customer will cause swing capacity to be exceeded at some time t, as follows:

```
(21) boolean client_fits(i ,t) {
        k = client.lookup(i).costOfService;
        if(client.lookup(i).contentSelection.averagePlayRate <=
                    swingCapacity(k,t) - server.cac_flowSafetyMargin)
              return true;
        else return false;
     }
```

System Description

Referring to FIG. **14**, a block diagram illustrating a preferred embodiment for a system implementing the methods presented herein is shown.

Block **1485** depicts plurality of network-based client computers that receive data from a server over the course of a session. Every client sessions has an associated session block **1435** within the server by means which a flow rate of content from client to server is effected and regulated to a given flow rate set-point. The flow optimizer **1430** manages bandwidth utilization across all sessions that have been admitted by the system call admission control (CAC) and the bandwidth call admission control blocks, labeled **1400**

and **1420** respectively. Specifically, the flow optimizer **1430** modulates the flow rate set-point of every active session so as to optimize aggregate flow or, more generally, cost.

Call Admission Control

Whenever a client contacts a server requesting service, the server must determine whether or not to admit the client. This admission function is performed in two separate phases. The system call admission control block **1400** considers whether or not sufficient internal resources exist to support the request. Toward this end, the system CAC **1400** estimates the needs of the prospective client with respect to a set of limited system resources such as I/O bandwidth, cache memory, open files, threads, etc. A request is denied if any required resource cannot be obtained. In the contrary case, the request is passed on to the Bandwidth CAC, which tests if the client fits and admits or rejects the client accordingly. The swing capacity is computed by the flow optimizer **1430** on a periodic basis according to equations (14) and (17) as appropriate. The managed bandwidth must reflect the true capacity of the attached network and the adapters leading thereto. This parameter should not be set arbitrarily high to prevent rejection of client requests. Doing so will cause the system to admit calls that cannot be properly handled, thereby diminishing the quality of the viewing experience.

Having admitted a client, block 1430 creates a client session block **1435**, which then begins to converse with its client-resident counterpart, **1480**) during their launch phase. Thereafter, the flow optimizer **1430** assumes control, and modulates flow over the session until either all content is delivered, or a failure to deliver is detected, or one of a number of VCR-like stream control events intervene (e.g. pause, rewind, fast-forward).

Server and Client Session Control Blocks

Data Transport Channel **1450** is responsible for transporting data packets originating from the flow regulator **1440** within the server to the former's peer entity **1460** within the client. The mode of communication between the peer entities may be connection-oriented (e.g. TCP, which delivers a byte stream in order, without duplication or loss) or datagram-oriented (e.g. UDP). Best-efforts datagram service is typically enhanced with acknowledgements to facilitate rapid recovery from occasional packet loss, out-of-sequence delivery, and duplication.

Peered session control channels **1445** and **1465** permit configuration and status data, together with miscellaneous commands, to be sent from client to server and vice-versa. A reliable connection-oriented transport service is typically employed for this purpose (e.g. TCP). Data and commands alike are multiplexed over this channel in the service of other entities within the clients or server. Thus, a datagram-oriented data transport channel **1460** within a client **1485** might employ the session control channel to ferry selective acknowledgements to its peer in the server. Similarly, the flow meter **1465** within the client **1485** forwards the measured flow rate to the associated flow regulator **1440** within the server.

The flow regulator **1440** obtains content data from a cache or disk, at the current offset into the content file, which it forwards as packets to the data transfer channel **1450**. The size and/or pacing of the packets are determined by the flow regulator **1440** in order to achieve the flow rate set-point imposed by the flow optimizer **1430**. The flow regulator **1440** is also responsible for determining the channel flow capacity to the client, which is the least of a configured value obtained from the client and a measured value. Toward this end, during the launch phase, the flow regulator **1440** might send content data packets to the client flow meter **1475** in a series of packet trains at progressively higher peak flow rates ending with the configured maximum flow rate, if any. The highest measured flow rate reported by the client flow meter **1475** is accepted as the flow rate capacity, which is posted for use by the flow optimizer **1430**. Subsequently, the flow regulator **1440** will compare the flow rate set-point with the flow rates reported by the flow meter **1465**, and down grade capacity as needed should the measured flow rate fall consistently fall below the requested flow rate. Whenever such a downgrading occurs, the flow regulator **1440** will test for subsequent relaxation of a transient constriction by means of the aforementioned series of packet trains in which the peak packet flow rate exceeds the delivered flow rate.

The buffer manager **1470** accepts data from the flow meter **1465**, which data is retained until it is consumed or passed over the content consumer **1480**. The buffer will typically be implemented as a ring buffer in memory or on disk. The write cursor will lag the read cursor by a no less than a specifiable time interval so as to permit rewinds of acceptable scope without requiring selective data re-retrieval from the server.

Flow Optimizer **1430**

The flow optimizer **1430** assumes control over a session once it has been launched, and its channel capacity has been determined. Subsequently, it modulates the flow rate set-point of every session such as to optimize aggregate flow or charges across all active sessions, subject to a variety of min/max constraints on sessions flows, and a maximum constraint on aggregate session flow, as previously discussed.

The flow optimizer **1430** continues to modulate session flow until all content has been delivered, or one of a number of session anomalies intervene, as follows:

A downgraded session flow capacity drops below the minimum session flow rate. The session is dropped by the optimizer **1430** as a solution involving this channel does not exist. The user is forced to apply for re-admission, with the option of accepting slower than real-time delivery at the present encoding rate, or accepting a degraded viewing experience at a reduced encoding rate better adapted to the presently available bandwidth.

A pause VCR command is received from the client. Flow to the client can continue until the client-side buffer is topped off, after which time flow drops to zero. The session is permanently dropped should the pause persists too long. Until that time, the CAC safety margin is temporarily increased by the jit (Just-In-Time) flow rate of the paused session. In this way the session can be resumed without the need for re-admission and the jit bandwidth is made available to currently sessions so long as the pause persists.

A VCR Jump-Forward command is received from the client. This action is problematic as it causes the just-in-time flow rate (calculated at the new forward offset) to increase, in violation of the assumptions of the CAC algorithm. It must be noted that this violation will occur even if the content that is sought has already been delivered to the client-side buffer. One workable policy is to force the client to apply for re-admission at the forward offset and with a suitably modified buffer level.

A VCR Jump-Backward (Rewind) command is received. This action may entail the need for re-transmission of content, as previously discussed. The just-in-time flow rate will decrease as per the new position and adjusted buffer contents.

Tunable Minimum Constraint

A parameterized family of minimum flow constraints can be defined as follows:

$$\text{wherein: } f_k(t)\text{=Target content flow rate for session k;} \tag{22}$$

$f_k^{JIT}(t)$=Just-in-time flow rate for session k; (RHS of Equation (5))

β=Burst tuning factor (positive or zero); and

$f_k^{min}(t)$=Minimum (reserved) flow rate for session k.

This family converges on $f_k(t)>=f_k^{JIT}(t)$ when β approaches 0. A β value of 0.1, for example, ensures a minimum flow rate that exceeds the JIT flow rate by 10%. Thus, non-zero values of β force early delivery of content to a player's media buffer, with beneficial effects on the viewing experience.

Alternative Call Admission Control

The traditional CAC algorithm compares the flow rate requirement of a prospective client with the unused bandwidth (obtained by subtracting the aggregate flow target from the server flow capacity) and grants service if unused bandwidth is not exceeded. This procedure may break down as the optimizer causes all available bandwidth to be used if it can, thereby reducing unused bandwidth to zero in the best case, even when a new client could be accommodated with a healthy margin to spare.

Nevertheless, a simple variation on this idea can be used. Available bandwidths can be defined as the difference between server flow capacity and the sum of minimum flow rates for each session computed according to (22) as follows:

$$f_{available}=f_{Svr}^{max}-(f_1^{min}+ \ldots +f_n^{min}) \tag{23}$$

wherein: $f_{available}$=Available bandwidth

f_{Svr}^{max}=Server aggregate flow capacity; and

f_n^{min}=Minimum (reserved) flow rate for session n.

Service is granted if the computed minimum flow rate of the prospective client is less than available bandwidth even when unused bandwidth is zero.

Toward a Tunable QOS

In the preceding discussion it is assumed that all clients share the same value of β. One might also consider a scheme whereby a client receives a β value according to the class/cost of service that client has signed up for: the greater the cost (and expected quality of service, or QOS) the greater the value of β.

Instead of maximizing charges by the methods described above, a variation on the algorithm to maximize flow is considered whereby the reserve flow computation for every session takes into account the individual β value of the associated client. In this fashion a single algorithm can serve either end. The optimizer first ensures that every session receives its required minimum bandwidth (which would now typically depend on β and elapsed session time) before allocating any excess bandwidth more or less evenly to all by interpolation. (According to FIG. **15**.)

The scheme is simpler computationally than the iterative interpolation method shown in FIG. **16**; it is also arguably more fair and stable with respect tot he allocation of available bandwidth.

Deadline-Driven Content Distribution

In delivering viewable or audible content to a player, the optimization algorithm imposes a minimum flow rate that ensures that all content is delivered by the implied deadline, which is the last moment of a play session. Notwithstanding, the optimizer attempts to deliver content earlier according to bandwidth availability within the server taken as a whole. A similar, though far simpler scenario arises relative to the distribution of content from an optimized server to a target device, whenever the delivery must be accomplished by a designated time, or equivalently, within a given elapsed time, and when overlapped non-blocking consumption of the content at a fixed rate is not required.

One example concerns the application where a session bundle comprises a session associated with the featured program together with sessions for every advertisement scheduled to interrupt the feature program at a designated time. Each session is endowed with a separate media buffer, which, in the case of ads, would typically be large enough to accommodate the ad in full, thereby permitting a seamless transition to the ad at the appointed time.

In adapting the optimizer to the needs of content distribution we must revisit the formulation of the flow constraints, as follows:

1. Media buffers are not of a concern, let alone buffer overflow, and it can be assumed that all content received is saved away in a file on disk, or in memory awaiting further use.

2. For these same reasons, the value of β may be zero, at which the minimum flow rate equals the just-in-time rate obtained by dividing the content as yet undelivered by the time till deadline:

$$f_k^{JIT}(t_k)=L_k^{TOGO}/(T_k-t_k) \tag{24}$$

wherein: L_k^{TOGO}=Content to go for session k, i.e., as yet undelivered by the server at time t, (client.lookup (k).session.payloadToGo).

Given the arbitrary format of the file to be distributed, T no longer signifies the "duration of play" at the average play rate (averagePlayRate); rather, T represents the elapsed time to the deadline measured at the moment the delivery session was launched, and t represents the elapsed time since session launch.

Session Bundles

A session bundle is a set of sessions that flow along a shared channel, and are thus subject to an aggregate flow constraint equal to the capacity of the shared channel. One important application of session bundles is connected with ad insertion.

It is not enough to treat each of the constituent sessions as independent with respect to the flow optimizer, for this practice might well result in session flow targets that do not exceed channel capacity when considered singly, yet exceed this same shared capacity when summed together.

A Solution to This Problem Involves Two Distinct Steps:

1. The sessions bundle is treated as a virtual session by the server optimizer, subject to the minimum flow constraint obtained by summation of the individual session minima, and a maximum constraint that is the least of (a) the sum of the least of constraints (3) and (4) for each session, and (b) the shared channel capacity client.lookup(k). channel.maxFlowRate (f_k^{cap}). The flow optimizer then generates an aggregate flow target for all channels virtual sessions by interpolation, using:

$$\text{Cost}(t)=f_1(t)+ \ldots +f_n(t) \tag{25}$$

The cost vector in this instance is the n-dimensional unit vector, where n is the number of active clients.

$$c=(1, 1, \ldots , 1) \tag{26}$$

Now it is possible to have an achievable aggregate flow target for the session bundle k that we must apportion

optimally among the constituent sessions. Fortunately for us, our interpolation procedure of FIG. **15** can be applied again to the sessions in our bundle, but where the aggregate flow target generated above replaces the aggregate flow constraint f_{Svr}^{max}

Specifically:

For every flow f_{ki}, within session bundle k, the optimal flow rate is obtained by interpolation between minimum and maximum session flows:

$$f_{ki}^{opt}=f_{ki}^{min}+\alpha^*(f_{ki}^{max}-f_{ki}^{min}) \quad (27)$$

$$\alpha=[(f_K^{bundle}-f_{k1}^{min}+ \ldots f_{kn}^{min})]/[(f_{k1}^{max}-f_{k2}^{min})++(F_{kn}^{max}-f_{kn}^{min})] \quad (28)$$

and the value of f_k^{bundle} is obtained from a higher level optimization in which session bundle k is treated as a virtual session for which:

$$f_k^{min}=f_{k1}^{min}+ \ldots +f_{kn}^{min} \quad (29)$$

$$f_k^{max}=\min(f_{k1}^{max}+ \ldots +f_{kn}^{max},f_k^{cap}) \quad (29)$$

It must be noted that session bundles are typically dynamic in nature, outliving many if not all of their constituent sessions. Before a session is added to the bundle two CAC decisions must be reached:

(1) establish bandwidth availability within the bundle; and

(2) bandwidth availability within the server.

Relationship to Adaptive Rate Control

The flow optimization methods should be viewed as constituting the lowest and perhaps only the first of many supervisory control layers to sit astride all streaming sessions, which comprise their own hierarchical stack of control functions; rate adaptive control sitting above a flow rate regulator which relies on a transport entity.

Channel capacity is known to vary widely over time. A capacity drop to below the minimum reserve flow rate is problematic as it causes the latter to rise in violation of its monotonicity assumption. In a server with available bandwidth to spare such a rise can be accommodated by the flow optimizer (by bumping the session's minimum flow reservation to the higher value) provided the flow deficit can be made up in the near term by draining pre-delivered content from the media buffer. Should either of these resources approach depletion, stream degradation to a lesser bit-rate or frame rate may be the only viable option short of a pause. Whether up or down, any rate change is a change to fe, which must be preceded by change to session's reserve flow rate allocation by the optimizer.

As can be seen, the optimizer and rate adapter must be integrated with care. One possible approach would be to modify a rate adapter such that it renegotiates its proposed fe value with the flow optimizer prior to effecting the rate change it deems appropriate based on information provided by the flow regulator. Rate increases might well be subject to deferral under heavy load conditions, awaiting bandwidth availability. The opposite is true of rate flow decreases: these would be forestalled as long as possible by the optimizer on a system that is not fully loaded, as previously discussed.

Live Events

Referring to FIG. **17**, the method of the present application can be extended to the optimization of "live", time-shifted "near-live" events, and cascaded delivery, wherein one server delivers content to another while the latter accepts clients for the same content. The extension assumes the following characteristics of the server and the live performance:

1. The performance is of a fixed duration T known in advance, encoded at an average rate fe, with a projected content payload of fe*T;

2. The origin server at the point of injection delays the stream by a small dwell time T^{DWELL} on the order of seconds (e.g. 10). Notwithstanding, the delayed content is available for early delivery to mirrored servers and players with the aim of providing these with a modicum of isolation from network jitter affecting the timely delivery of the live video frames.

3. Throughout the live event, captured content is stored in a file, thereby permitting late joiners to view the "live event" starting from any point between the present and the event start time;

4. At the conclusion of the live event, the encoded content, now fully captured to a file, is available for viewing over the balance of the admission window (e.g., 60 minutes in FIG. **17**).

The optimisation of live events and video on demand (VOD) differ principally in one respect, namely in the formulation of the server content underflow constraint.

For VOD, this constraint is

L_k^{TOGO}(t) is initialised to the content size (fe*T) at that start of play, and (31)

decremented as content is delivered to the client. Δt is the time interval between successive periodic optimizations (e.g. 1 second). In practice this constraint rides well above other operative maximum flow constraints (e.g. buffer overflow, channel capacity) until the very last calculation of the session flow target by the optimiser.

For a live stream, this constraint must additionally reflect the fact that content is streaming into the server (where it builds up), even as the server streams it out toward clients, and that the latter activity cannot possibly overtake the former.

Referring to FIG. **17**, (which illustrates how optimised live streams behave) line L**1** depicts the build-up of freshly captured content to the server. Line L**2**, which is delayed (time-shifted) relative to L**1** by the dwell time T^{DWELL}, represents the live performance as cast and viewable by clients. One such client is admitted at time A and joins the live session at the live moment B. The dashed curved line linking points B and C shows a plausible build-up of content in the client's media buffer in advance of play, as the viewer continues its advance along line L**2**. Past point C the top of the media buffer tracks the live capture stream and thenceforth advances along line L**1**. In this instance, the time shift between the live performance as cast (L**2**) and the live performance as captured (L**1**) imposes a maximum on the amount of content that could ever accumulate in a client media buffer, irrespective of its actual capacity.

Another client, admitted at time t=10, is interested in viewing the live stream from the beginning without missing anything. As before, the dashed line joining 10 and D depicts a plausible build-up of content in the client's media buffer in advance of play, as the viewer continues its advance along line L**3**, which is time-shifted by (10+T^{DWELL}) minutes relative to L**1**. The greater time shift (now between L**1** and L**3**) permits more extensive pre-buffering of "near live" content.

The present constraints do not impose suitable flow limits at and beyond points C and D when the client delivery streams finally links up with the live capture stream, in the live moment. Consequently the optimiser may propose flow targets that cannot be realized.

Accordingly, the following modification for use in connection with live or cascaded streams is made:

$$f_k(t) <= [(L_k^{TOGO}(t) - Live^{TOGO}(t)]/\Delta t \quad (32)$$

for a live event delivered at flow rate fe, we have

$$\text{Live}^{TOGO}(t) = \max(fe*(T - T^{DWELL} - t), 0) \quad (33)$$

$\text{Live}^{TOGO}(t)$ is the span between projected content size fe*T (level line **L0** in FIG. **17**) and the content capture level line **L1**, whereas $L_k^{TOGO}(t)$ is the gap between **L0** (i.e., fe*T) and the dashed curved line representing the content level delivered to the media buffer. The latter must always exceed the former. As the two approach one another this constraint takes effect, causing $L_k^{TOGO}(t)$ to always exceed $\text{Live}^{TOGO}(t)$ as it decreases at rate fe.

Viewers joining late and willing to see the live performance with some delay relative to the live case enjoy higher potential levels of isolation from network disturbances, and thus QOS, by virtue of their greater effective buffer capacity. By the same token, such viewers afford the optimizer greater latitude to optimize the flow of VOD and "live" content across load fluctuations, thereby enhancing server capacity. Viewers that wish to experience the live event in "real-time" nevertheless benefit by a modestly enhanced QOS conferred by a dwell time's worth of content in their buffers.

SUMMARY

A method and system for call/connection acceptance and flow modulation for network delivery of video/audio programming is thus provided. Although several embodiments have been illustrated and described, it will be apparent to those skilled in the art that various changes and modifications may be made without departing from the spirit of the invention or the scope of the claims.

What is claimed is:

1. A method of bandwidth allocation for delivery of stored digital content from at least one server device to at least one client device by way of a network, the method comprising the steps of:

a) prescribing a control variable which represents a target flow rate of content from the server device to each client device by:
   i) forming a hyperplane of control variables that aggregate to a maximum allowed aggregate flow rate;
   ii) computing a first vector of minimum allowed client flow rates;
   iii) computing a second vector of maximum allowed client flow rates;
   iv) determining a multi-dimensional shape of which an interior diagonal spans the first and second vectors; and
   v) finding a solution of control variables at a point of intersection between the diagonal and the hyperplane;

b) determining time-varying constraints on the target flow rate of the content;

c) determining a cost function of the control variables for all clients wherein the cost function represents an aggregate flow rate and is a sum of all flow rates for all clients; and

d) prescribing bandwidth to all clients based upon a value of the control variables that maximize the cost function.

2. The method of claim **1** wherein step (d) comprises performing periodic computations to update the value of the control variable such that the bandwidth can be continuously allocated to each client.

3. The method of claim **2** wherein a new client is accepted by:

i) determining an admission capacity of the bandwidth;

ii) admitting a prospective client if the clients minimum allowed value of the control variable is less than the admission capacity; and

iii) wherein a client admitted for service is guaranteed to have sufficient content flow over the entire session.

4. The method of claim **3** wherein:

the admission capacity equals a server swing capacity reduced by a predetermined safety margin which may be zero, the swing capacity equaling a difference between a server flow capacity and a sum of a minimum allowed flow rate for all clients.

5. The method of claim **4** wherein the minimum allowed value of the control variable is initialized to an average consumption rate.

6. The method of claim **4** wherein the minimum allowed value of the control variable is initialized to the average consumption rate multiplied by a factor greater than one.

7. The method of claim **4** wherein the server flow capacity varies in a step-wise function over time according to a predetermined schedule and the method of accepting a new client further comprises:

i) determining a sequence of future step changes of server flow capacity;

ii) determining at each time in the sequence of future step changes a value for a future worst swing case capacity, the worst case swing capacity being obtained from the server flow capacity at the time by subtracting an extrapolated present minimum allowed flow rate for all active clients that are potentially active in the future;

iii) admitting a prospective client if a minimum allowed value of the control variable is less than a capacity obtained from the sequence and a present capacity;

iv) whereby a new client can be admitted without compromising the service to existing clients in the presence of previously scheduled server flow capacity changes and all clients accumulate content in their buffers to the extent of their respective buffer capacity.

8. The method according to claim **7** wherein the method of accepting a new client further comprises:

i) partitioning available bandwidth into partitions representing a prescribed service charge;

ii) determining a maximum-minimum capacity for a service charge to be the maximum allowed sum of all minimum allowed flow rates for all clients incurring an equal or lesser service charge, the maximum-minimum capacity equaling the sum of bandwidth allocations for all service charges less than or equal to the service charge;

iii) determining a service-charge-specific swing capacity for a given service charge as the difference between the maximum-minimum capacity for a service charge and the sum of all minimum allowed flow rates for all clients incurring the service charge or a lesser service charge;

iv) determining the server swing capacity to be the server flow capacity minus a margin and minus the sum of the minimum allowed flow rate for all clients; and

v) determining a service-charge-specific capacity to be the least of the server swing capacity and the charge specific swing capacity.

9. The method of claim **8** wherein the method of accepting a new client further comprises:

i) determining the sequence of future times of step changes to server flow capacity over a proposed consumption period;

ii) determining for each time in the sequence of future times, a future worst-case value for a service-charge-specific capacity, the worst case value being obtained by using the extrapolated present minimum allowed flow rate of all presently active clients that are potentially active at the time in the future; and

iii) admitting a client incurring a given service charge if the minimum flow rate of the client is less than the capacity obtained from a sequence and a present service-charge-specific capacity.

10. The method of claim **3** wherein step (b) comprises determining time-varying constraints wherein:

i) an aggregate flow rate for all clients does not exceed a predetermined server flow capacity;

ii) a flow rate from the server to the client does not exceed a maximum allowed flow rate for the client;

iii) a flow rate to the client will never cause a buffer of the client to overflow;

iv) a flow rate to the client stops when the content is exhausted; and

v) a flow rate from the sever will never be less than the client's minimum allowed flow rate which may vary over time and may be zero.

11. The method of claim **10** wherein the maximum allowed flow rate to the client is given by the minimum of one selected from the following:

i) a client flow rate ceiling;

ii) a flow rate required to fill the client buffer before the next periodic computation; and

iii) a flow rate required to complete the delivery of the content before the next periodic computation.

12. The method of claim **11** wherein:

i) the minimum allowed client flow rate cannot increase over time, but may decrease such that an initial value of the control rate is no less than an average consumption rate;

ii) the control variable for the client is always greater than or equal to the current minimum allowed client flow rate; and

iii) the client buffer will never underflow until the content is fully consumed at the average consumption rate.

13. The method of claim **12** wherein the minimum client flow rate is a product of a constant content flow rate from the client to the server that causes a last piece of content to be delivered at the last possible moment as the client buffer is being drained at the average consumption rate, with a factor greater or equal to one.

14. The method of claim **13** wherein the minimum client flow rate is initialized to the average consumption rate.

15. The method of claim **13** wherein the minimum client flow rate is initialized to the average consumption rate multiplied by a factor greater than one.

16. The method of claim **13** further comprising the step of calculating the minimum client flow rate periodically.

17. The method of claim **1** wherein the cost function represents a service charge and is the sum of the client flow rates multiplied by the client's cost of service.

18. The method of claim **17** wherein:

step (a) comprises:

i) determining a maximum allowed flow rate and a minimum allowed flow rate for each client;

ii) determining a flow rate range for each client as the difference between the maximum allowed flow rate and the minimum allowed flow rate; and

iii) initializing a current flow rate for each client as the minimum allowed flow rate and summing the flow rate into the total server flow rate; and

step (d) comprises:

i) computing remaining server bandwidth as the difference between the maximum server flow capacity and the total server flow rate; and

ii) allocating remaining server bandwidth to remaining clients wherein all clients with a given cost of service receive bandwidth equally to the extent of their respective ranges, and no client with a given cost of service being allocated bandwidth unless all clients with a higher cost of service have received their maximum possible allocation of bandwidth.

19. The method of claim **1** wherein step (v) comprises finding a solution of control variables at the second vector if there is no point of intersection between the diagonal and the hyperplane.

20. The method of claim **1** wherein step (a) comprises:

i) computing the sum of the minimum allowed flow rates to all active clients;

ii) computing the sum of the flow rate ranges for all active clients;

iii) computing the difference between the maximum allowed aggregate flow rate and the sum of the minimum;

iv) computing a factor as the ratio of the difference over the sum of the ranges;

v) if the factor exceeds 1, the control variable is set to the maximum allowed client flow rate; and

vi) if the factor is less than 1, computing the control variable by multiplying the client range by the factor and then adding the result to the minimum allowed client flow rate.

21. The method of claim **1** wherein:

step (a) comprises:

i) determining the maximum allowed flow rate and minimum allowed flow rate for each client;

ii) determining the flow rate range for each client as the difference between the maximum flow rate and the minimum flow rate; and

iii) initializing the control variable for each client as said minimum allowed flow rate and summing the flow rate into a total server flow rate; and

step (d) comprises:

i) computing remaining server bandwidth as the difference between the maximum server flow capacity and the total server flow rate; and

ii) allocating remaining server bandwidth to remaining clients to the extent of their respective ranges.

22. The method of claim **21** wherein the step of allocating remaining server bandwidth further comprises the clients receiving the bandwidth equally to the extent of their respective ranges.

23. The method of claim **1** wherein the maximized value of the control variable can be determined by:

i) determining a maximum and minimum flow rate for each client;

ii) determining a range between the maximum and minimum flow rates in order to find a flow rate range;

iii) determining available bandwidth by finding the difference between an aggregate flow capacity and a sum of the minimum flow rate;

iv) determining a flow variable by dividing the unused bandwidth by the flow rate range;

v) prescribing the control variable to be the minimum flow rate added to the flow rate range if a flow variable is less than one or prescribing the control variable to be the minimum flow rate added to the flow rate range corrected by a flow variable if the flow variable is greater than one such that the control variable is easily calculated for each client.

24. The method of claim **1** wherein at least one session is a session bundle and:

1) the minimum flow constraint is computed as the sum of the minimum flow rate constraints for all bundled sessions;

2) the maximum flow constraint is the least of 1) the sum of the maximum individual flow rate constraints of each session and 2) the flow capacity of the channel shared among the bundled sessions; and

3) the target flow rate is computed for the session is apportioned among bundled constituent sessions.

25. The method of claim **24** wherein the bundled session is an advertisement.

26. A system for allocating bandwidth between a server device and at least one client device, the system comprising:

a call acceptance module operative to receive an incoming request for service;

a flow regulator configured to deliver content at a modulated target flow rate, the content being delivered between the server device and a respective client device when a call is accepted by the call acceptance module; and

a flow optimizer configured to modulate the target flow rate of the flow regulator in order to optimize an aggregate flow of content by performing the following procedure:

a) prescribing a control variable which represents a target flow rate of content from the server device to each client device by

i) forming a hyperplane of control variables that aggregate to a maximum allowed aggregate flow rate;

ii) computing a first vector of minimum allowed client flow rates;

iii) computing a second vector of maximum allowed client flow rates;

iv) determining a multi-dimensional shape of which an interior diagonal spans the first and second vectors; and

v) finding a solution of control variables at a point of intersection between the diagonal and the hyperplane;

b) determining time-varying constraints on the target flow rate of the content;

c) determining a cost function of the control variables for all clients wherein the cost function represents an aggregate flow rate and is a sum of all flow rates for all clients; and

d) prescribing bandwidth to all clients based upon a value of the control variables that maximize the cost function.

27. The system of claim **26** wherein the flow optimizer is configured to modulate the flow rate of the flow regulator in order to optimize charges for the content.

28. The system of claim **26** wherein the flow optimizer is configured to determine a control variable which corresponds to an optimized flow rate.

29. The system of claim **28** wherein the flow optimizer is configured to generate a cost function of the control variable that corresponds to a maximized value of the control variable.

30. The system of claim **26** wherein the flow regulator is configured to deliver the modulated flow rate of content in response to the control variable.

31. The system of claim **30** wherein the call acceptance module is configured to accept a call based upon a value of the control variable.

32. The system of claim **26** wherein the flow optimizer is configured to determine a maximized value of the control variable by:

i) determining a maximum and minimum flow rate for each client;

ii) determining a range between the maximum and minimum flow rates in order to find a flow rate range;

iii) determining unused bandwidth by finding the difference between an aggregate flow capacity and a sum of the minimum flow rate;

iv) determining a flow variable by dividing the unused bandwidth by the flow rate range;

v) prescribing the control variable to be the minimum flow rate added to the flow rate range if the flow variable is less than one or prescribing the control variable to be the minimum flow rate added to the flow rate range corrected by the flow variable if the flow favorable is greater than one such that the control variable is easily calculated for each client.

33. A method of bandwidth allocation for delivery of stored digital content from at least one server device to at least one client device by way of a network, the method comprising the steps of:

a) prescribing a control variable which represents a target flow rate from the server to each client device based upon the amount of buffer of the client by:

i) forming a hyperplane of control variables that aggregate to a maximum allowed aggregate flow rate;

ii) computing a first vector of minimum allowed client flow rates;

iii) computing a second vector of maximum allowed client flow rates;

iv) determining a multi-dimensional shape of which an interior diagonal spans the first and second vectors; and

v) finding a solution of control variables at a point of intersection between the diagonal and the hyperplane;

b) determining time-varying constraints on the flow rate of the content;

c) determining a cost function of the control variables for all clients in response to a size of the client's buffer wherein the cost function represents an aggregate flow rate and is a sum of all flow rates for all clients; and

d) prescribing bandwidth to all clients based upon a value of the control variables that maximize the cost function.

* * * * *

EXHIBIT G

US007334044B1

(12) **United States Patent**
Allen

(10) **Patent No.: US 7,334,044 B1**
(45) **Date of Patent: *Feb. 19, 2008**

(54) **METHOD FOR CONNECTION ACCEPTANCE CONTROL AND OPTIMAL MULTI-MEDIA CONTENT DELIVERY OVER NETWORKS**

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(73) Assignee: **Burst.com**, Santa Rosa, CA (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

This patent is subject to a terminal disclaimer.

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(22) Filed: **Jun. 25, 1999**

Related U.S. Application Data

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(51) **Int. Cl.**
G06F 15/16 (2006.01)

(52) **U.S. Cl.** **709/233**; 709/223; 709/224; 709/231; 709/232; 709/230; 709/219; 370/229; 370/233; 370/236; 370/252; 370/468

(58) **Field of Classification Search** 709/223, 709/226, 231, 203, 232, 233, 234; 370/352, 370/229–238, 254–258, 386, 400, 401, 422, 370/428–429, 477, 486–490, 468; 725/105–106, 725/114, 118
See application file for complete search history.

(56) **References Cited**

U.S. PATENT DOCUMENTS

5,502,816 A * 3/1996 Gawlick et al. 709/227
5,504,744 A 4/1996 Adams et al.
5,583,995 A * 12/1996 Gardner et al. 709/219
5,633,810 A 5/1997 Mandal et al.
5,838,663 A 11/1998 Elwalid et al.
5,838,921 A 11/1998 Speeter
5,978,357 A 11/1999 Charny
5,982,748 A * 11/1999 Yin et al. 370/232
5,995,488 A 11/1999 Kalkunte et al.
6,047,328 A 4/2000 Charny et al.
6,052,384 A * 4/2000 Huang et al. 370/468

(Continued)

OTHER PUBLICATIONS

Kim, S. et al., "Call Admission Control for Prioritized Adaptive Multimedia Services in Wireless/Mobil Networks", *VTC 2000-Spring. 2000 IEEE 51st. Vehicular Technology Conference Proceedings*. Tokyo, Japan, May 15-18, 2000, *IEEE Vehicular Technology Conference*, New York, NY., (May 15, 2000) IEEE, US, vol. 2 of 3: Conf. 51, pp. 1536-1540.

(Continued)

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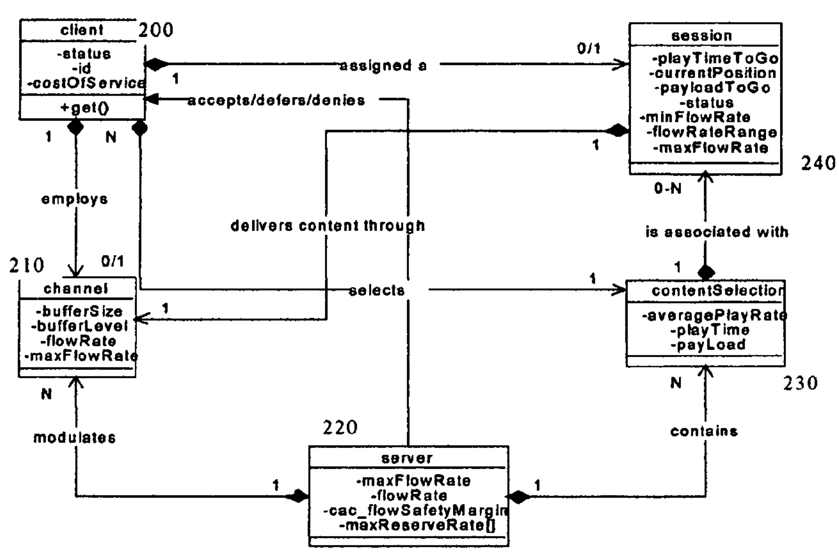
(74) *Attorney, Agent, or Firm*—Buchanan Ingersoll & Rooney PC

(57) **ABSTRACT**

A method is disclosed for call and/or connection acceptance control and the optimal delivery of multimedia (audio/video) data over networks. This method involves the establishment and monitoring of certain criteria which may be used to maximize the number of simultaneous clients without sacrificing quality-of-service for already-connected clients. Methods are disclosed for maximizing total throughput as well as maximum charge models for different levels of service. The disclosed methods solve these optimization problems by expanding on linear-program techniques in manners geared towards multimedia content delivery over networks and many variations suitable for varying business models are disclosed.

19 Claims, 12 Drawing Sheets



U.S. PATENT DOCUMENTS

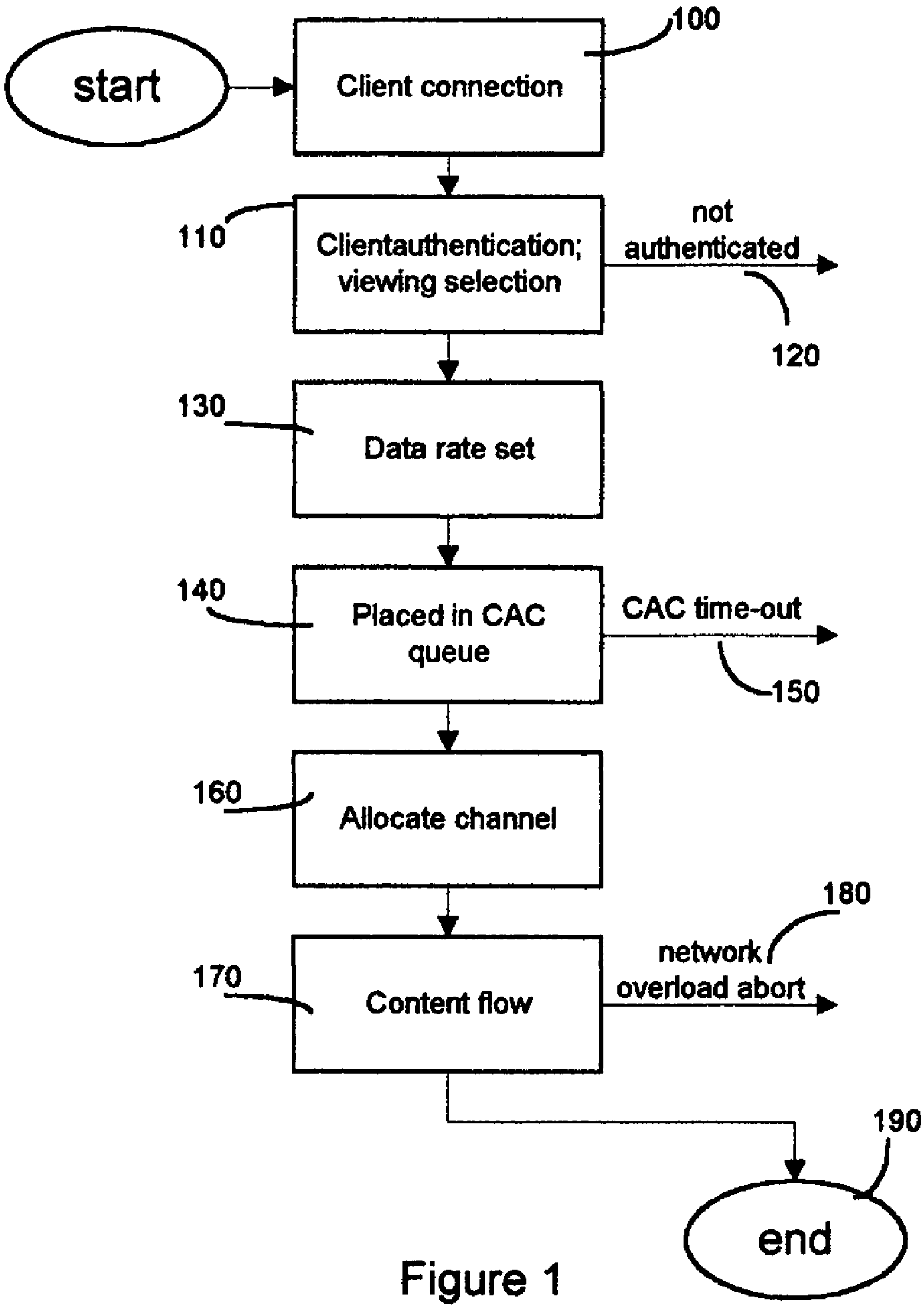
6,069,894	A *	5/2000	Holender et al.	370/397
6,125,396	A	9/2000	Lowe	
6,192,029	B1	2/2001	Averbuch et al.	
6,212,169	B1 *	4/2001	Bawa et al.	370/252
6,240,103	B1 *	5/2001	Schoenblum et al.	370/468
6,295,294	B1 *	9/2001	Odlyzko	370/389
6,304,551	B1 *	10/2001	Ramamurthy et al.	370/232
6,331,986	B1 *	12/2001	Mitra et al.	370/468
6,400,686	B1	6/2002	Ghanwani et al.	
6,466,979	B1 *	10/2002	Plouffe, Jr.	709/226
6,493,317	B1	12/2002	Ma	
6,597,662	B1	7/2003	Kumar et al.	
6,647,419	B1	11/2003	Mogul	
6,721,789	B1	4/2004	DeMoney	
6,850,965	B2 *	2/2005	Allen	709/203
6,950,399	B1	9/2005	Bushmitch et al.	
7,027,403	B2 *	4/2006	Porikli et al.	370/238
7,191,246	B2 *	3/2007	Deshpande	709/233
2003/0016664	A1	1/2003	MeLampy et al.	
2003/0072327	A1	4/2003	Fodor et al.	
2004/0090974	A1	5/2004	Balakrishnan et al.	
2004/0136379	A1	7/2004	Liao et al.	
2006/0165010	A1	7/2006	Christiance et al.	
2006/0176806	A1	8/2006	Yoshihra et al.	
2006/0224768	A1	10/2006	Allen	

OTHER PUBLICATIONS

Reininger, D. et al., "A Dynamic Quality of Service Framework for Video in Broadband Networks", *IEEE Network, IEEE Inc.,* New York, US (Nov. 6, 1998), 12:6, pp. 22-34.

Beard, C. et al., "Connection Admission Control for Differentiating Priority Traffic on Public Networks", *Military Communications Conference Proceedings, 1999, Milcom 1999. IEEE* Atlanic City, NJ, USA Oct. 31-Nov. 3, 1999, Piscataway, NJ, USA, *IEEE*, US (Oct. 31, 1999), pp. 1401-1408.

* cited by examiner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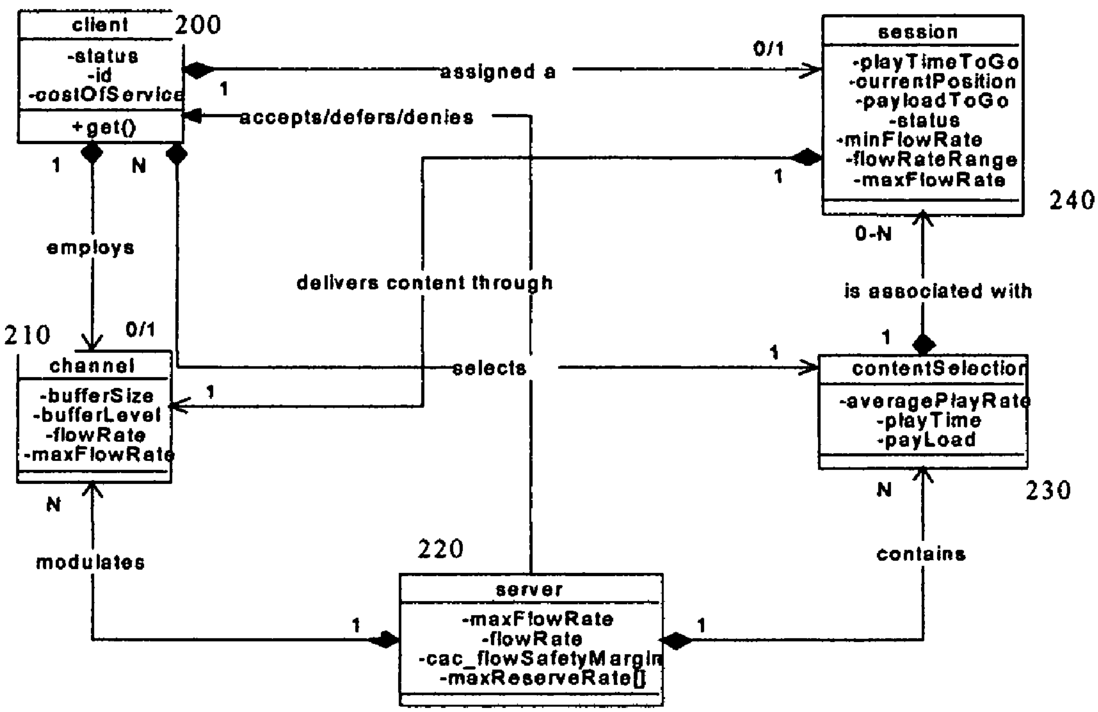
start
Client connection
110
Clientauthentication; viewing selection
not authenticated
120
130
Data rate set
140
Placed in CAC queue
CAC time-out
150
160
Allocate channel
180
network overload abort
170
Content flow
190
end

Figure 1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200
-status
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-costOfService
+get()
assigned a
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accepts/defers/denies
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-payloadToGo
-status
-minFlowRate
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240
1
N
1
employs
delivers content through
0-N
is associated with
210
0/1
channel
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-bufferLevel
-flowRate
-maxFlowRate
selects
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1
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-playTime
-payLoad
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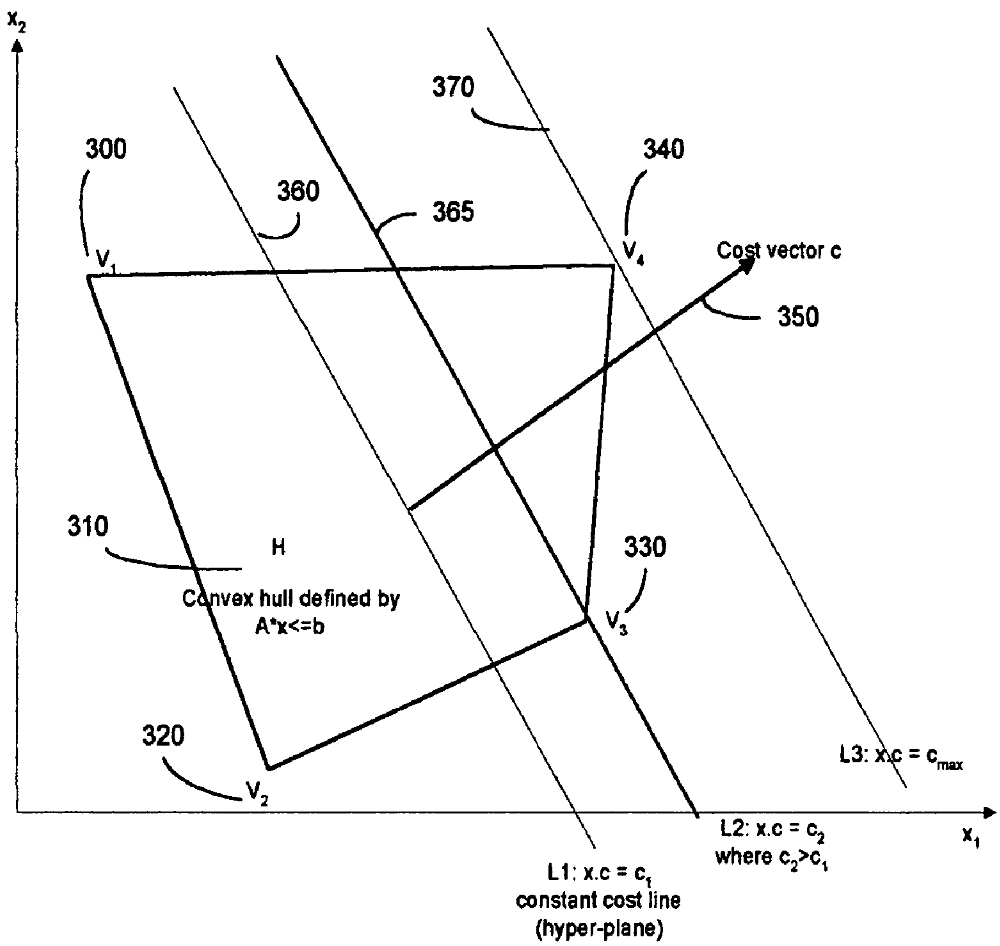
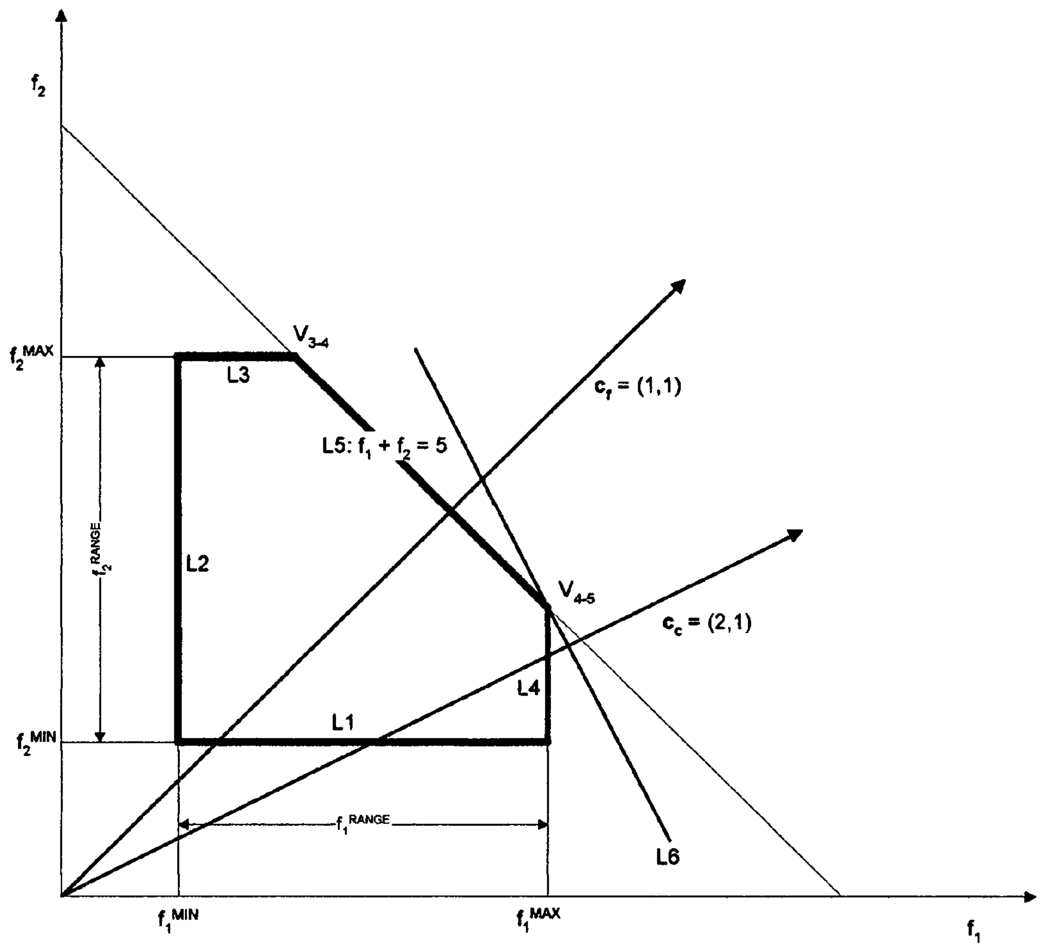
1

Figure 2



x_2
370
300
340
360
365
V_1
V_4
Cost vector c
350
310
H
Convex hull defined by
A*x<=b
330
V_3
320
V_2
L3: x.c = c_{max}
L2: x.c = c_2
where c_2>c_1
x_1
L1: x.c = c_1
constant cost line
(hyper-plane)

Figure 3



f_2
f_2^{MAX}
$V_{3\text{-}4}$
L3
$c_f = (1,1)$
L5: $f_1 + f_2 = 5$
f_2^{RANGE}
L2
$V_{4\text{-}5}$
$c_c = (2,1)$
L4
L1
f_2^{MIN}
f_1^{RANGE}
L6
f_1^{MIN}
f_1^{MAX}
f_1

Figure 4

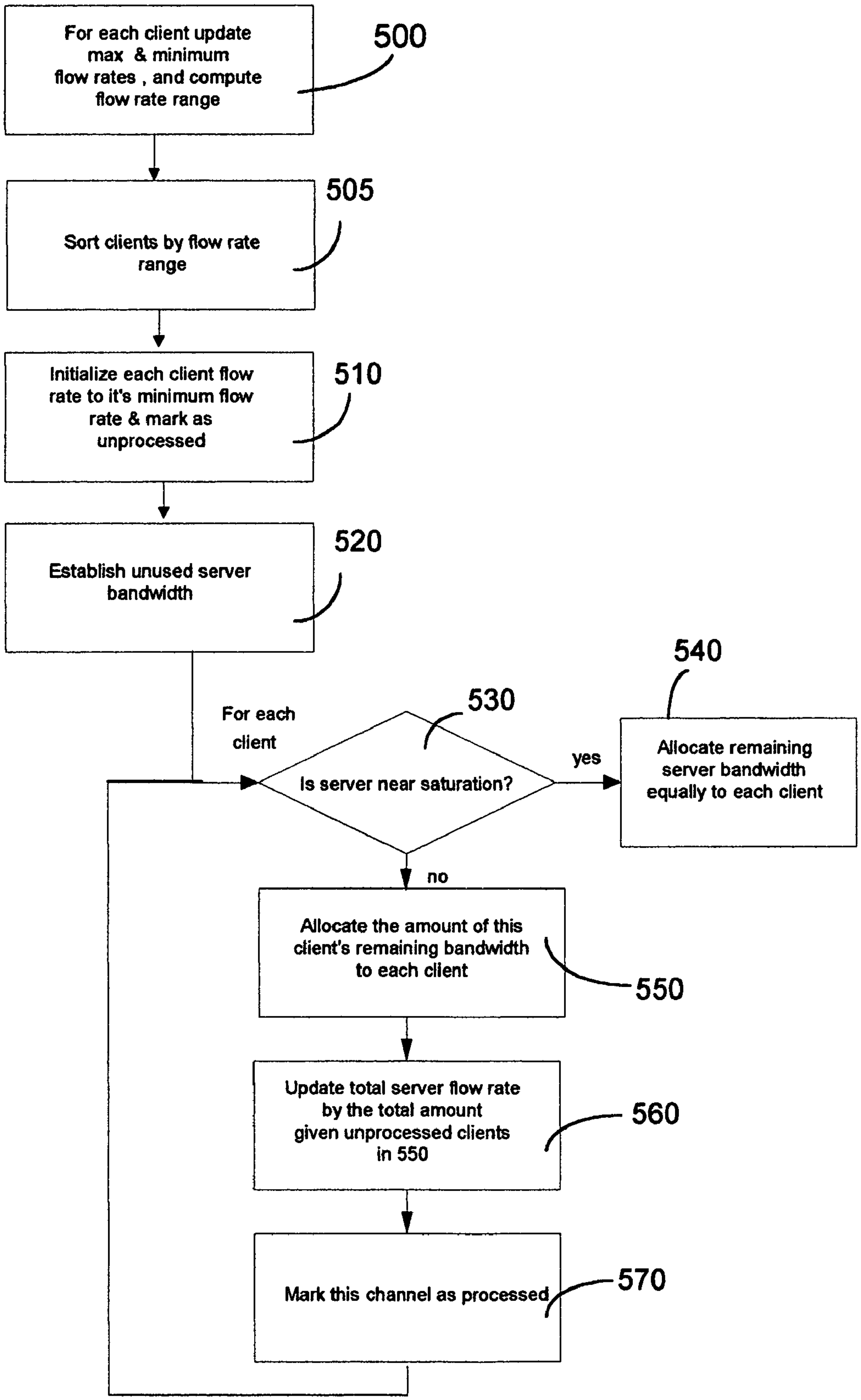
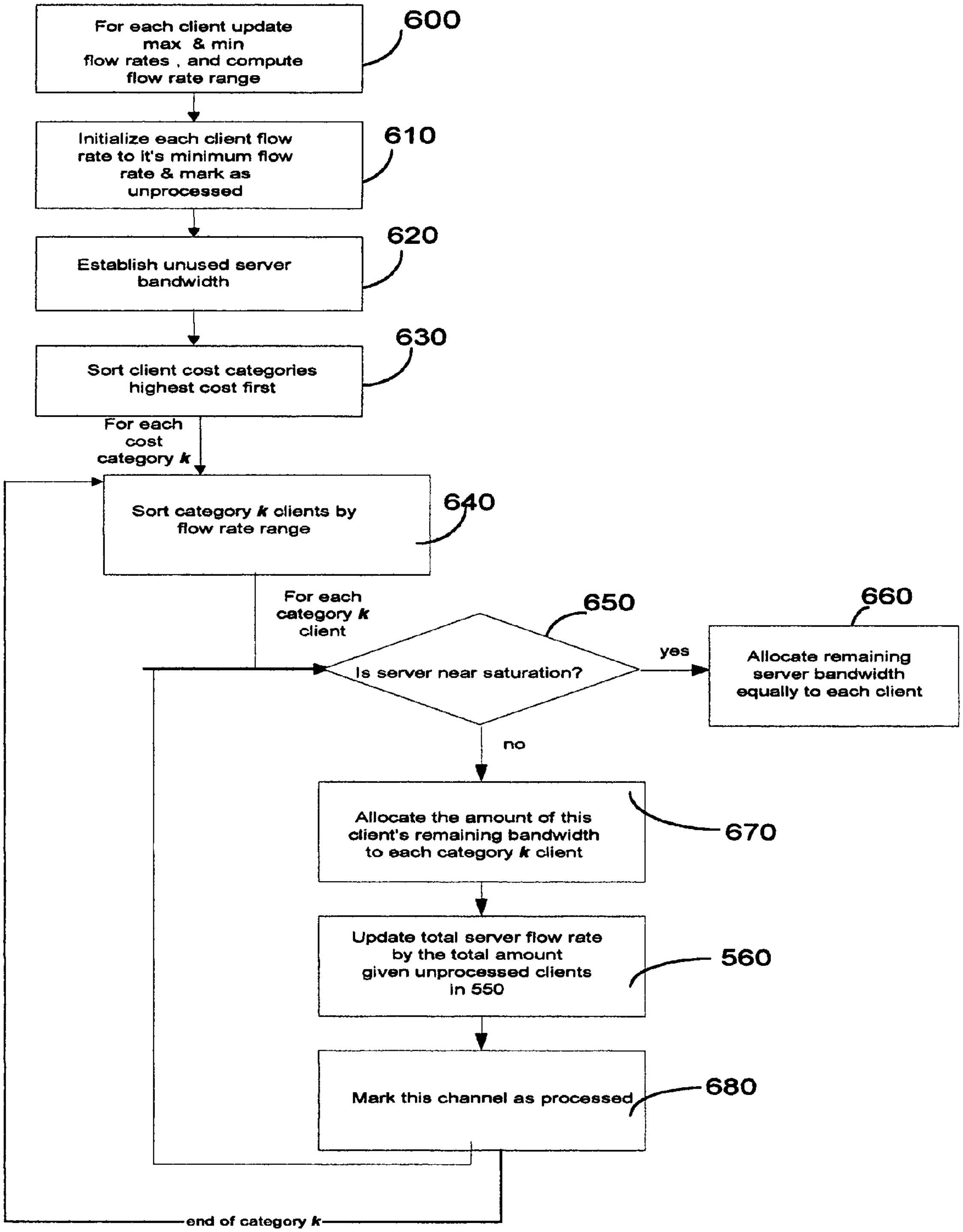

For each client update max & minimum flow rates , and compute flow rate range
500
Sort clients by flow rate range
505
Initialize each client flow rate to it's minimum flow rate & mark as unprocessed
510
Establish unused server bandwidth
520
For each client
Is server near saturation?
530
yes
Allocate remaining server bandwidth equally to each client
540
no
Allocate the amount of this client's remaining bandwidth to each client
550
Update total server flow rate by the total amount given unprocessed clients in 550
560
Mark this channel as processed
570

Figure 5



For each client update max & min flow rates , and compute flow rate range
600
Initialize each client flow rate to it's minimum flow rate & mark as unprocessed
610
Establish unused server bandwidth
620
Sort client cost categories highest cost first
630
For each cost category *k*
Sort category *k* clients by flow rate range
640
For each category *k* client
Is server near saturation?
650
yes
Allocate remaining server bandwidth equally to each client
660
no
Allocate the amount of this client's remaining bandwidth to each category *k* client
670
Update total server flow rate by the total amount given unprocessed clients in 550
560
Mark this channel as processed
680
end of category *k*

Figure 6

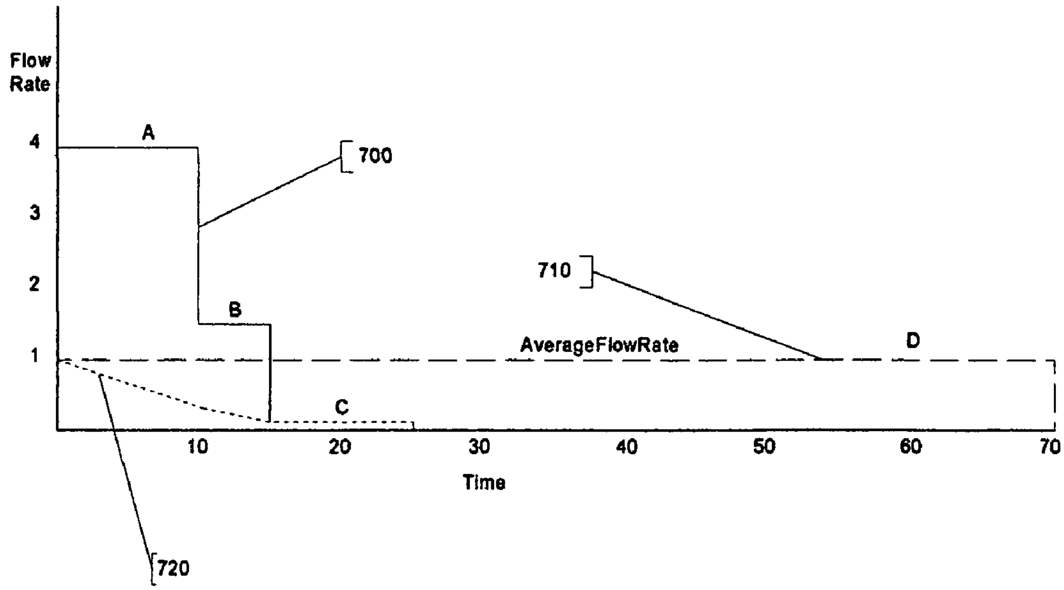


Figure 7

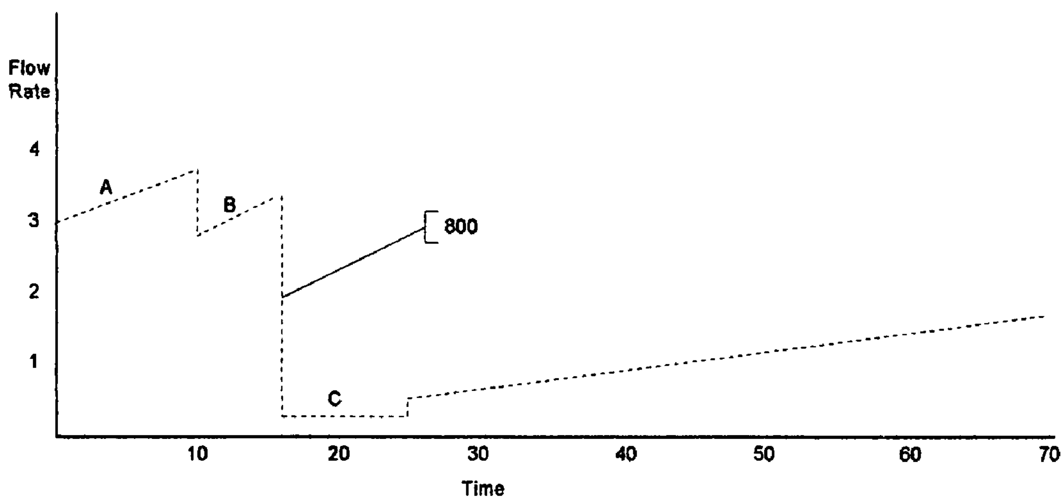


Figure 8

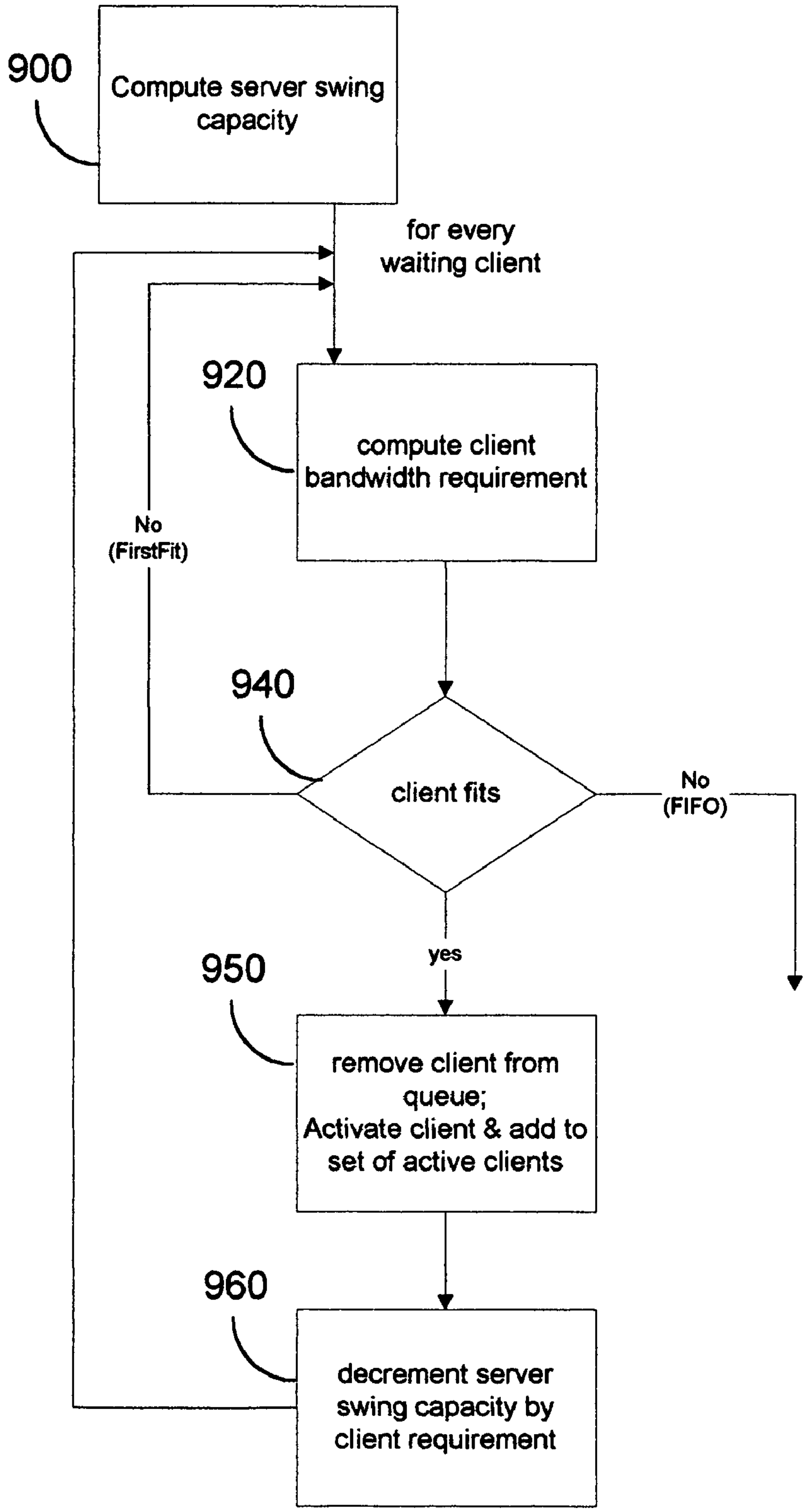

900
Compute server swing capacity
for every waiting client
920
compute client bandwidth requirement
No (FirstFit)
940
client fits
No (FIFO)
yes
950
remove client from queue; Activate client & add to set of active clients
960
decrement server swing capacity by client requirement

Figure 9

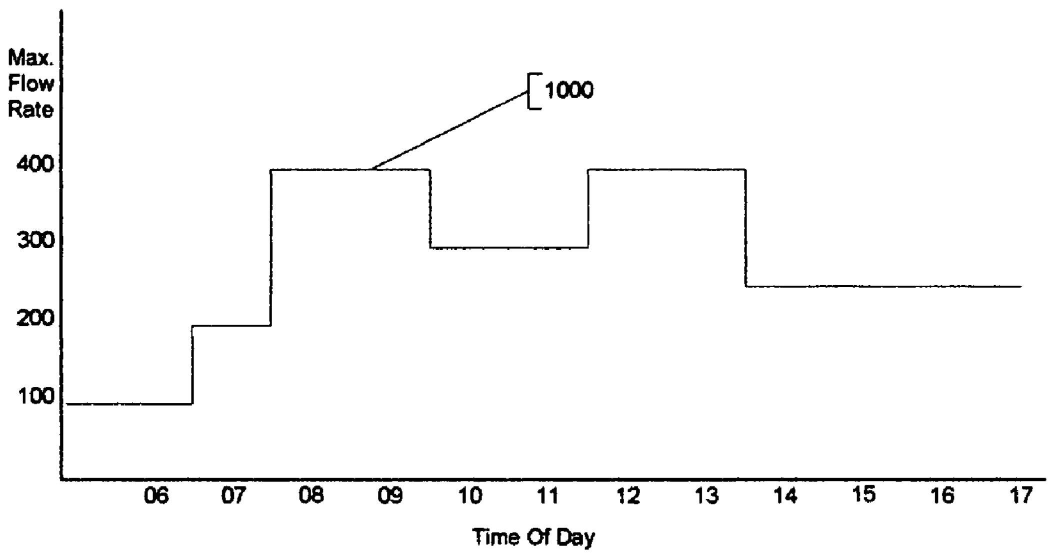

Max. Flow Rate
400
300
200
100
1000
06
07
08
09
10
11
12
13
14
15
16
17
Time Of Day

Figure 10

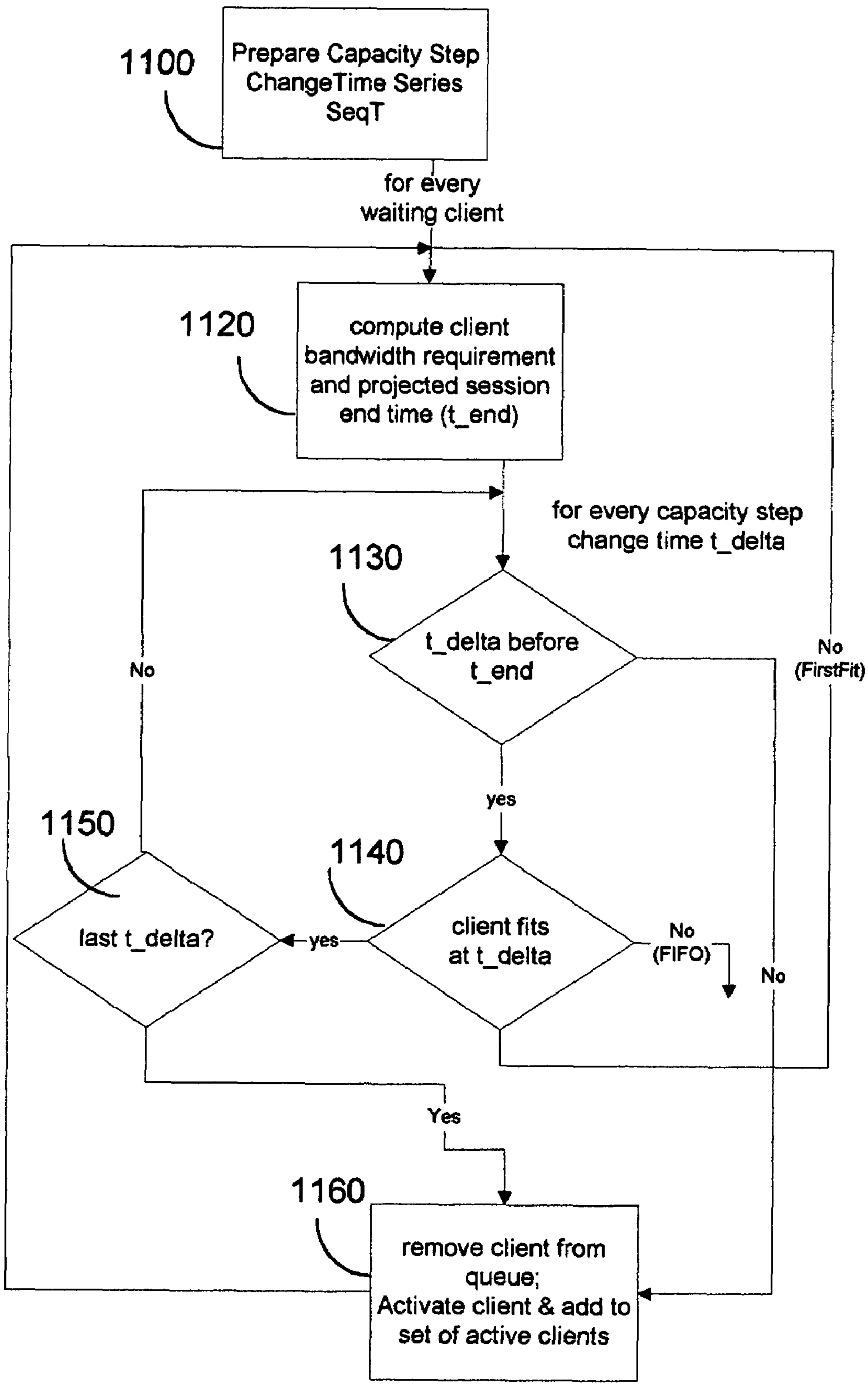

1100
Prepare Capacity Step ChangeTime Series SeqT
for every waiting client
1120
compute client bandwidth requirement and projected session end time (t_end)
for every capacity step change time t_delta
1130
t_delta before t_end
No (FirstFit)
No
yes
1150
1140
last t_delta?
yes
client fits at t_delta
No (FIFO)
No
Yes
1160
remove client from queue; Activate client & add to set of active clients

Figure 11

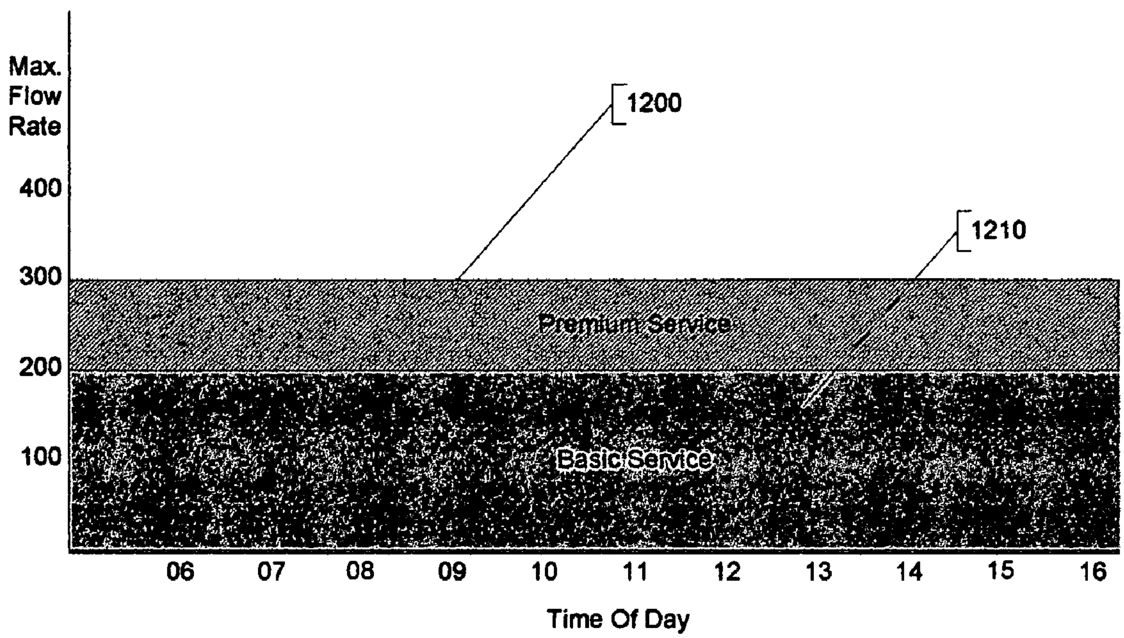

Max. Flow Rate
400
300
200
100
1200
1210
Premium Service
Basic Service
06
07
08
09
10
11
12
13
14
15
16
Time Of Day

Figure 12

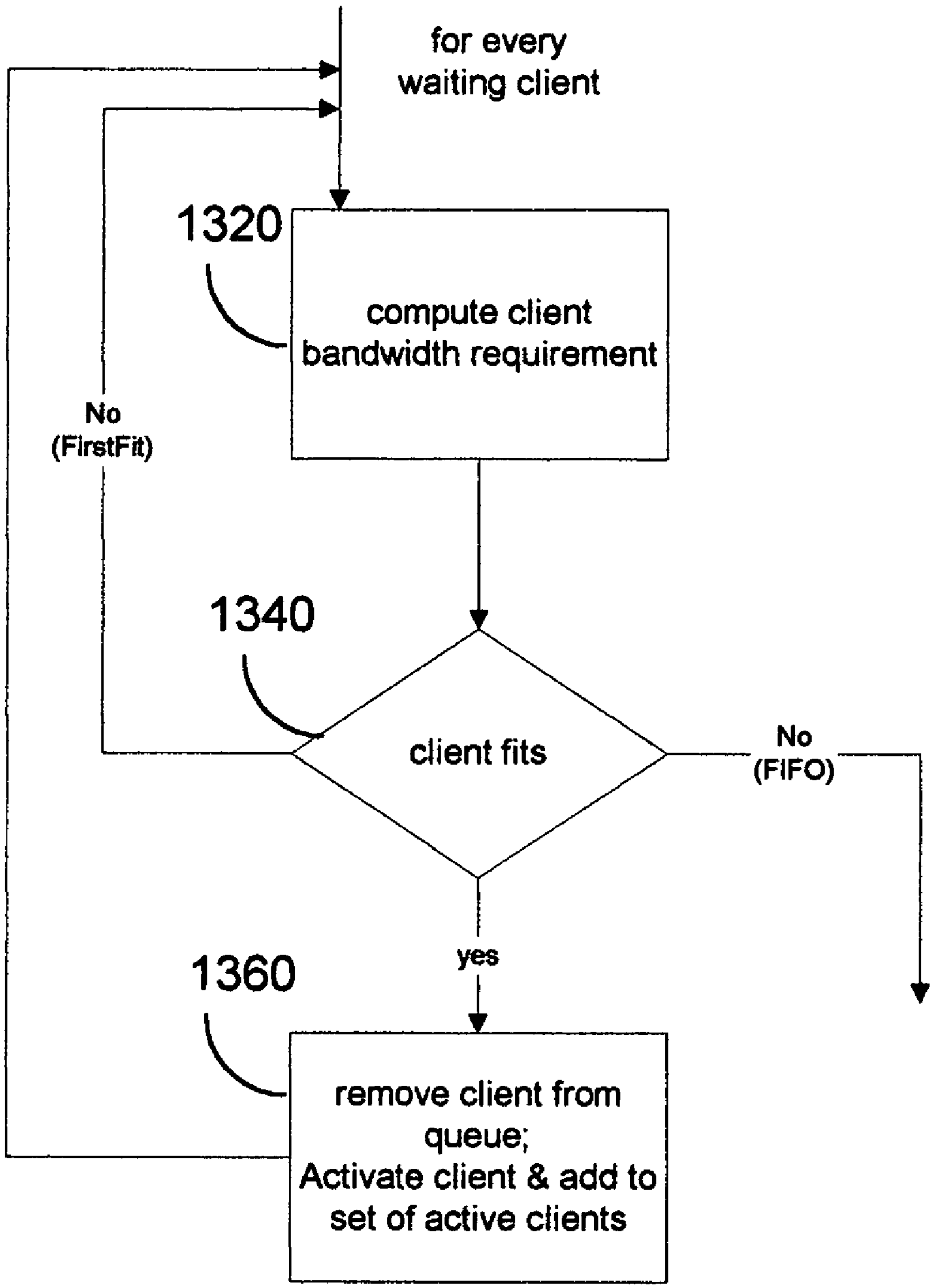

for every
waiting client
1320
compute client
bandwidth requirement
No
(FirstFit)
1340
client fits
No
(FIFO)
yes
1360
remove client from
queue;
Activate client & add to
set of active clients

Figure 13

METHOD FOR CONNECTION ACCEPTANCE CONTROL AND OPTIMAL MULTI-MEDIA CONTENT DELIVERY OVER NETWORKS

CROSS-REFERENCE TO RELATED APPLICATIONS

This invention claims the priority date of provisional application No. 60/108,777, "Method for Connection Acceptance Control and Optimal Multimedia Content Delivery Over Networks", inventor Arthur Allen, filed Nov. 17, 1998.

BACKGROUND OF THE INVENTION

1. Field of the Invention

The invention relates to the field of delivery of multimedia content over a variety of networks. More specifically, it pertains to multimedia servers which service many clients simultaneously for the delivery of multimedia content which is used and played back at each client. It addresses methods for determining optimal delivery rates to each client and methods for determining whether new clients may be accepted without diminishing the quality of service to existing clients.

2. Description of Related Art

In the history of multimedia program delivery, some in the industry have long advocated the use of large client-side buffers and faster-than-real-time content delivery over a network as offering the best of all worlds: a jitter-free viewing experience and a cost-effective utilization of the network resources at hand. Few systems, however, go very far in addressing how to schedule clients or a method for accepting new clients. Real-time systems, often known as streaming systems, can schedule new clients in a very simple manner—if sufficient bandwidth remains for the added real-time stream, then the client may be accepted. However, such systems do not maximize the number of simultaneous clients. On the other hand, faster than real-time delivery, sometimes known as store-and-forward systems, open up the possibility for more flexible scheduling procedures to control and optimize the number of simultaneous clients while ensuring a high level of quality of service.

The methods for such call acceptance and flow modulation that have been proposed in the prior art have been largely ad-hoc and also incomplete. These have been ad-hoc in the sense that there has been no guiding rationale for their selection from among many possible and potentially superior alternatives. The methods have also been incomplete insofar as they did not address the question of whether any given incoming request for service should be accepted or denied. Video-on-demand systems, or more generally, any system in which a multimedia server is designed to serve multiple clients over a network to deliver bounded content, can benefit from the use of such flow modulation techniques and call acceptance procedures.

BRIEF SUMMARY OF THE INVENTION

Optimal Content Flow Modulation

The present invention addresses multimedia content delivery optimization by re-casting the problem to be solved as an optimization problem in which one seeks to maximize a designated value function moment-by-moment, subject to a set of real-world operational constraints which will typically vary over time. Accordingly, given a set of clients and

associated sessions, an optimal delivery procedure continuously establishes content flow rates from the content server to each of its clients so as to maximize aggregate value according to the governing value function.

This approach holds several advantages: 1) optimization problems are well understood, and are tractable by a large and diverse collection of computational methods; 2) if it exists, the global solution that is obtained is arguably optimal by construction, and thus superior or equal to all other.

The present invention teaches the method of optimizing two particular value functions:

1) total data delivered (maximize throughput).

2) total delivery charges (maximize charges).

The first value function does not distinguish one customer from another and will deliver as much data as possible from server to clients irrespective of the characteristics of the latter. The second value function favors the service of high paying customers. It can easily be seen that the first function is a special case of the second one whereby all clients are charged equally.

As will be seen in this disclosure, optimizing for these functions and identifying the necessary constraints requires a new and unique perspective that is specifically designed for the multimedia environment. Moreover, the disclosed methods are specifically designed to account for and accommodate real-world scenarios of today's networks. Consequently several variations of the method are presented to accommodate various scenarios.

The following briefly-defined concepts are useful in understanding the present invention:

Call/Connection Acceptance Control (CAC)

In accordance with the invention, a CAC procedure is responsible for deciding whether a candidate for service can be accommodated without jeopardizing sessions already in progress at the present time or at some time in the future; failing that, it must decide whether a service request should be queued for a time, or rejected.

Flow Modulation

Flow modulation methods are those portions of the system which manage the communication and data flow between the server and the clients. Collectively, these methods provide the multimedia data to the client and provide the server with the information about the state of the transmission, playback, user status and network status. These parameters are further used by the present invention in the CAC procedures. In fact, as will be shown, the proposed CAC procedures are tightly integrated with the flow modulation methods.

Adaptation to Variations in Network Capacity

Operational constraints may change over time. For instance, one might elect to vary the total bandwidth available for multimedia content delivery according to the time of day. Alternatively, exogenous data flows on the network may cause unexpected disturbances by usurping available bandwidth and impeding the delivery of data along established session channels. The content delivery strategy of the present invention includes the ability to adapt to scheduled as well as unexpected disturbances so as to minimize unwanted disruptions of services.

Burst Transmissions Provide the Opportunity to Adapt

The present invention, due to it's faster-than-realtime transmissions (also know as burst transmissions), which are realized by use of high-bandwidth networks and large client cache or intermediate storage, provides an opportunity to adapt to changing network conditions. In contrast, real-time (streaming) systems of the prior art are essentially designed

for worst-case scenarios: each client must be assumed to constantly use the complete real-time playback bandwidth. Such prior art systems are unable to adapt to any derivation from this scenario. For example, take the simple case where the total server bandwidth is 100% utilized by all clients playing back the streaming video. Should any network condition change, such as a temporary decrease in available bandwidth over the network, then one or more clients' playback is interrupted, and the system can not recover from such a condition until the bandwidth is regained. Even worse, if a single client presses pause either that unused bandwidth must remain reserved and no more clients can be accepted, or that paused client is pushed out in order to service the new client. In essence little or no CAC procedure may be implemented.

In contrast, the present invention burst transmits portions of a program and immediately 'gets ahead of itself', thus allowing headroom for a myriad of methods to intelligently handle new clients, client interactivity and possible network fluctuations.

In accordance with the invention, methods are disclosed for optimally determining the flow rate to each client. Methods are also disclosed for accepting or rejecting new clients; these call-acceptance methods are tightly coupled with said flow rate modulation methods. A series of constraint expressions are presented which govern the methods for determining the flow rates and acceptance of new clients. Linear programming techniques are used to optimally solve these expressions. Various embodiments are presented including scenarios for multiple-rate tariffs, and time-of-day bandwidth variations.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. **1** depicts the flow of control and/or data between the different stations of a content delivery session;

FIG. **2** illustrates the Entity Data Model;

FIG. **3** geometrically illustrates the problem statements;

FIG. **4** geometrically illustrates an expansion of the problem statement;

FIG. **5** illustrates a method for implementing flow modulation;

FIG. **6** illustrates a method for implementing flow modulation for maximized charges;

FIG. **7** illustrates typical content flow;

FIG. **8** illustrates typical server swing capacity;

FIG. **9** illustrates a method for call-acceptance and control (CAC);

FIG. **10** illustrates planned constraints on maximum flow;

FIG. **11** illustrates a method for call-acceptance and control (CAC) with scheduled flow changes;

FIG. **12** illustrates stratification of services; and

FIG. **13** illustrates a method for call-acceptance and control (CAC) for maximal charge.

DETAILED DESCRIPTION OF THE INVENTION

FIG. **1** depicts the flow of control and/or data between the different stations of a content delivery session in accordance with the invention. As shown, a client attempts a connection (**100**) and manifests itself to the Content Selection subsystem by means of a low bandwidth control channel (not shown). Next the client is authenticated and a selection is made (**110**), typically with the aid of a browser software such as Netscape™ or Microsoft Internet Explorer™. If the client is not authenticated, it is dismissed from the system

(**120**). If the client has been authenticated and a program selected for viewing then the rate of service is set at this point (**130**), perhaps according to the selection that was made, or some contractual stipulation. The client is now placed on the service queue of the CAC subsystem (**140**). A client that is made to wait too long will eventually balk (**150**). Assuming this does not occur, the CAC subsystem will eventually allocate a channel to the client and open a session (**160**). Control now devolves upon the Content Flow Modulator (not shown) which starts the flow of content from server to client (**170**). Subsequent capacity changes, whether predictable or not, may force in abrupt termination of a session In progress (**180**). Otherwise the session runs to completion (**190**).

The entities entering into our discussion are depicted in FIG. **2**. Client **200** maintains certain data associated with this entity; as shown but not labeled, which includes without limitation, status, id and costOfService. The other entities also each include unlabeled but depicted data. The diagram farther depicts the relationship between each entity. As shown, client **200** is assigned a session **240**. Client **200** employs a channel **210**. Client **200** selects contentSelection **230**. Session **240** delivers content through channel **210**. Server **220** modulates channel **210**. Server **200** contains contentSelection **210**. Server **220** accepts, defers or denies client **200**. And contentSelection **230** is associated with session **240**.

Furthermore FIG. **2** depicts the various one-to-many relationships. Each client **200** employs one channel **210**. Client **200** may or may not receive one of channel **210**, as depicted by the 0/1 notation. Similarly, client **200** may or may not receive a session **240**. However, whenever client **200** does receive a session **240**, it will always receive a channel **210** since channel **210** and session **240** are allocated as a pair. One or more (N) of client **200** may select one of contentSelection **230**. And server **220** contains one or more (N) of contentSelection **230**. Each one of contentSelection **230** is associated with 0-N of session **240**. Each session **240** delivers content through one of channel **210**. And server **220** modulates one or more (N) of channel **210**.

A more detailed list of each entity of FIG. **2**, and each one's associated description, data elements and function calls is listed below. This listing closely resembles that of object-oriented programming. As such, 'methods' represent the ability to obtain or modify data, while 'attributes' represent data which is directly associated with that particular entity. The listing also includes information relating to one embodiment wherein software programming specifics are disclosed, such as a variable type (double, int and so forth) and more. The present invention is not limited to such an embodiment and other implementations are possible without deviating from the scope and intent of the present invention. The listing, however detailed, is merely illustrative of the data and functions which are used in the equations and methods described herein.

Consequently, data and functions from this listing, associated with the various entities, will be used in forthcoming equations, flowcharts and methods. The reader is directed to this listing as reference when reading such equations and examining such drawings.

—start of entity data model detailed listing—

Model: Untitled 1 (public)

Contains:
client, session, channel, server, contentSelection.

Component: client (public Class/Interface)

Comment:

A client entity stands for a client presently requesting or receiving service.

Methods:

public static lookup (id: in int): client
public GetId ( ): const int&
public SetId (val: in int&)
public GetCostOfService ( ): const double&
public SetCostOfService (val: in double&)

Attributes:

private status: client<int>

Specifies whether or not a client has been allocated a channel and session.

private id: int

Integer-valued identifier that is unique to the client (primary key). Can be obtained from a monotonically increasing counter.

private costOfService: double

Dollar charge per Mbyte. This value is the same for all customers under flow optimization. Under cost/charge optimization may be an integer value reflective of the rank; the higher the rank the higher the charge.

Has:

public selected: contentSelection
public assigned a: session
public employs: channel

Component: session (public Class/Interface)

Comment:

A session entity holds various state information about the service being received by an associated customer.

public GetCurrentPosition ( ): const double&
public SetCurrentPosition (val: in double&)
public GetPayloadToGo ( ): const double&
public SetPayloadToGo (val: in double&)
public GetStatus ( ): const int&
public SetStatus (val: in int&)
public GetMinimumFlowRate ( ): const double&
public SetMinimumFlowRate (val: in double&)
public GetFlowRateRange ( ): const double&
public SetFlowRateRange (val: in double&)
public GetMaxFlowRate ( ): const double&
public SetMaxFlowRate (val: in double&)

Attributes:

private playTimeToGo: double

Indicates the minutes remaining in the viewing experience. Initialized to contentSelectiont.playTime (see below).

private currentPosition: double

Pointer into media content from which content is being delivered.

private payloadToGo: double

The amount of media content (in Mbytes) as yet undelivered by the server.

Does not include any content presently stored in the client-side buffer.

private status: int

Indicates whether session is active or paused.

private minimumFlowRate: double

This is the minimum flow from server to client required to ensure uninterrupted service over the remaining playTime. Has a value of zero if playloadToGo is zero. Given by (payloadToGo*8)/(playTimeToGo*60)

private flowRateRange: double

Specifies the effective range over which the channel content flow serving a session is constrained without consideration for interactions with other flows. Equals maxFlowRate−minimumFlowRate

private maxFlowRate: double

Effective maximum bound on flow as expressed in formula (8) which must be re-evaluated periodically.

Has:

public delivers content through: channel

Component: channel (public Class/Interface)

Comment:

A channel represents the network resources from server to client associated with an ongoing session, encompassing the client-side buffer if any, and its level.

public GetBufferLevel ( ): const double&
public SetBufferLevel (val: in double&)
public GetFlowRate ( ): const double&
public SetFlowRate (val: in double&)
public GetMaxFlowRate ( ): const double&
public SetMaxFlowRate (val: in double&)

Attributes:

private bufferSize: double

Capacity of the client-side buffer (or equivalent).

private bufferLevel: double

Current buffer level in MBytes of stored content.

private flowRate: double

Flow rate through channel specified by the relevant optimizing flow modulator.

private maxFlowRate: double

This value represents the maximum possible flow rate from the server to an individual client over its "channel". This value reflects restrictions on flow that pertain to an individual client. It may be determined by factors such as the bandwidth of client's link to the network, or a limit imposed administratively to ensure balanced network utilization.

Component: server (public Class/Interface)

Comment:

Entity representing the media server and its CAC and flow modulation activities.

public GetFlowRate ( ): const double&
public SetFlowRate (val: in double&)
public GetMaxMinFlowRate[ ] ( ): const double&
public SetMaxMinFlowRate[ ] (val: in double&)

Attributes:

private maxFlowRate: double

Maximum possible content flow that is allocated to the server by the network.

private FlowRate: double

Aggregate content flow rate, summed over all sessions and their associated channels.

private cac_flowSafetyMargin: double

Tunable safety margin used by the CAC algorithm to protect sessions-in-progress from being affected by changes in available network bandwidth.

private maxMinFlowRate[ ]: double

Applies when N rate tariffs exist. This array holds the maximum floor level for each category of service. The value for the costliest category N is stored in maxMinFlowRate[N−1], and for the least costliest in maxMinFlowRate[0]. It is the relative magnitude of these ascending values that matters, not their absolute value. Thus the actual maximum floor flow rate for category

k is given by server.maxFlowRate*(server.maxMinFlowRatc[k−1]/server.maxMinFlowRate[N−1]). Similarly, the maximum floor flow rate for category N is server.maxFlowRate.

Has:
public contains: contentSelection
public modulates: channel

Component: contentSelection (public Class/Interface)

Comment:
Entity represents a video/sound clip or other bounded unit of content. A continuous data feed does not qualify.

Attributes:
private averagePlayRate: double
The average rate at which media content is consumed by the client, as computed by dividing the (payload*8) by the (playTime*60)
private playTime: double
Duration of play of the media content in minutes.
private payLoad: double
total size of the content in Mbytes.

Has:
public is associated with: session

—end of entity data model detailed listing—

The following table summarizes the highlights of the previous detailed description of each entity in FIG. **2**.

Entity	Description
client 200	Each client is denoted by an associated unique integer index_{Id}. The set of active clients is denoted by $S_{\text{activeClients}}$. The set of deferred clients is denoted by $S_{\text{QdClients}}$. Incoming clients are expected to select the content they wish to view prior to being queued for dispatch by the CAC sub-system, which requires knowledge of the client's bandwidth requirements, duration of play, and cost of service, all of which may vary according to the selection.
server 220	Servers sit astride a network and can deliver media content through the network to their clients up to a designated maximum flow rate. The server is responsible for accepting or rejecting clients, launching sessions and associated channels for the former, and modulates content flows over all channels in an optimal manner.
channel 210	A channel represents the data path between the server and the client. The channel buffer is typically located near or within the client's viewing station. The flow of content through the channel is set by the flow modulator sub-system.
contentSelection 230	A server will typically act as a repository for media content, which it can deliver to clients upon demand. For purposes of the invention, media content is characterized by its payload and the play duration, which together imply the averagePlayRate = (payload*8)/(playTime*60). The averagePlayRate is the streaming rate imposed by realtime just-in-time streaming algorithms.
session 240	Every session represents an instance of media content delivery to an associated client over a designated channel. The playTimeToGo indicates the time remaining before the content is fully played out to the client. The payloadToGo is the amount of content data as yet undelivered to the channel. A session terminates when this value reaches zero, at which time playTimeToGo may still be large, according to the capacity, the level of the channel buffer, and the media play rate.

Below are presented some formulas and problem statements which are used in the methods which follow.

The flow of content between entities is subject to the following constraints at all times. Buffer levels are always expressed in Mbytes and data rates in Mbits/sec.

$\Sigma_{i \in S_{activeClients}}$(client.lookup($i$).channel.flowRate) <=server.maxFlowRate The sum of all channel flows cannot exceed the imposed maximum throughput capacity of the server. (1)

client.lookup(i).channel.flowRate<=client.lookup(i).channel.maxFlowRate for all $i \in S_{activeClients}$ (2)

The data path from server to client is subject to its own constriction.

$$(3)\ client.lookup\ (i).channel.flowRate <= (client.lookup\ (i).channel.bufferSize - client.lookup\ (i).channel.bufferLevel) * 8/60 + client.lookup\ (i).session.mediaContent.averagePlayRate,\ \text{for all } i \in S_{activeClients},$$

The channel buffer is never allowed to overflow.

client.lookup(i).channel.flowRate<=client.lookup(i).session.payloadToGo*8/60 for all $i \in S_{activeClients}$, (4)

Content that does not exist can not be delivered. (Constraint I will ordinarily prevail except at the very end of a session.)

The constraints listed above are straightforward applications relating to the flow of data through constricted channels, out of finite data sources, and into and out of bounded buffers. By contrast, the following constraint, which imposes a minimum channel flow rate instead of a maximum, is less obvious. The minimum value, termed the minFlowRate is set to the flow rate which, if sustained over the balance of the play time to go (playTimeToGo), ensures that all required content will be available when needed—and no sooner—until all content is played out. This floor value can be calculated for $i \in S_{activeClients}$ by the formula

client.lookup(i).session minFlowRate=(client.lookup(i).session.payloadToGo*8)/(client. lookup(i).session.playTimeToGo*60) (5)

Thus:

client.lookup(i).channel.flowRate>=client.lookup(i).session.minFlowRate for all $i \in S_{activeClients}$ (6)

The variable constraint bounds (i.e. the values to the right of the inequality symbol) of equations 1-4 and 6 are reevaluated on a periodic basis (e.g. once per second) prior to the execution of the CAC procedure and optimizer. In particular, the minFlowRate value starts out at the beginning of a session equal to the streaming rate. By construction the minFlowRate rate never exceeds this initial value so long as constraint 6 is honored. In fact, constraint 5 implies that the min[f]FlowRate value must be a diminishing function of time that may hold its value for a time but never rises. As seen from equation 6, the actual data rate of the channel, flowRate, is always greater than or equal to the minFlowRate. By design, and virtue of the fact the present invention uses faster-than-realtime transmissions, the system quickly gets ahead of itself and ensures that after initial conditions, the minFlowRate is always equal to or less than the real-time rate and that it continues to decrease. As we shall see the

CAC procedure exploits this monotonic characteristic of the minimum flow rate over time.

Constraints 2, 3 and 4 are of like kind, each specifying an upper bound on individual channel flows. Whereas the bound for constraint 2 is typically a constant, the bounds on 3 and 4 will vary over time. Nevertheless, only one of the three bounds is effective at any given time, namely the one with the smallest bound value, given by:

client.lookup(*i*).session.maxFlowRate=minimum of

1) client.lookup(i).channel.maxFlowRate,

2) (client.lookup(i).channel.bufferSize−

client.lookup(i).channel.bufferLevel)*8/60+

client.lookup(i).session.mediaContent.average-playRate,

3) client.lookup(i).session.payloadToGo*8/60 (7)

Consequently, formulas 2, 3, and 4 can be consolidated into a single constraint, the bound for which is computed at every scan to be the smallest bound of associated constraints 2, 3 and 4.

client.lookup(*i*).channel.flowRate<=client.lookup(*i*).session.maxFlowRate, whereby for all $i \in S_{activeClients}$, maxflowRate is given by equation (7). (8)

At any one time, individual channel flows are constrained over a range, as follows:

client.lookup(*i*).session.flowRateRange=client.lookup(*i*).session.maxFlowRate−client.lookup(*i*).session.minimumFlowRate (9)

Value Functions

The value functions introduced in the Description of Related Art can now be expressed mathematically as linear functions of channel flows as follows:

Optimizing Throughput (Maximal Flow)

value=$\Sigma i \in S_{activeClients}$client.lookup(*i*).channel.flowRate) (10)

Optimizing Charges (Maximal Charges)

value=$\Sigma i \in S_{activeClients}$(client.lookup(*i*).channel.flowRate)*client. lookup(*i*).costOfService) (11)

Optimization Problem

The optimization problem, which in one embodiment is strictly throughput and in another case is charge, can be stated simply as follows:

Find values for

client.lookup(i).channel.flowRate for all $i \in S_{activeClients}$ constrained by inequalities 1 through 5, such that the value obtained by evaluating expression 10 or 11 assumes a maximum.

Both of these problem formulations are examples of Linear Programming for which a number of well-known and generally effective computational solutions exist. In linear programming one seeks to optimize a linear cost function of variable x

$$c*x=c_1*x_1+\ldots+c_n*x_n \quad (12)$$

subject to a set of linear inequality constraints

$$A*x<=b \quad (13)$$

where $x^T=(x_1, x_n)$, $c=(c_1, \ldots, c_n)$ are the state variable & cost vectors, A is an n-by-m matrix, $b^T=(b_i, \ldots b_m)$ is the constraint vector, and the operator '*' stands for matrix or scalar multiplication.

FIG. **3** is introduced as illustrative of the problem statement and the general methods of the prior art, and is not incorporated as an element of the invention.

The linear programming problem as well as its solution can best be understood with the aid of geometry. FIG. **3**, depicting a 2-dimensional Cartesian problem space, inequality constraints (13) define a convex hull H **310** over which a search for an optimum value of $x=(x_1,x_2)$ is permitted to range. The cost vector **c350** defines an infinite family of equal cost lines (hyperplanes) which lie orthogonal to c. Three examples of such lines are shown in L_1**360**, L_2**365**, and L_3**370**, each of progressively higher value. The supreme value of the cost function is obtained by sliding along c **350** till one can go no farther, in this instance toward vertex V_4**340** of hull H **310**. Many well-known methods (e.g. the Simplex Method) work roughly in this fashion, exploiting the fact that at least one optimum point must be at a vertex. In particular, the Simplex method algorithm begins by finding a vertex (e.g. V_2**320**), and then moves along a sequence of vertices (e.g. V_3**330**, V_4**340**) improving itself each time until no further improvement is possible & the summit is reached.

Let us suppose instead that V_3**330** were placed along L_3**370** along with V_4**340**. According to prior art methods, V_3**330** and V_4**340** are the two possible solutions, but the equally valuable points in between them are not. As we shall soon see, the problem of throughput optimization (6) falls in this category.

While vertex V_1**300** does not factor into this description, it is depicted in FIG. **3** for completeness.

Flow Modulation

A Method for Maximal Flow

The following relates to one embodiment for optimizing the total data flow.

FIG. **4** depicts a scenario involving two flows. The convex hull is in this instance bounded by line segments L**1**, L**2**, L**3**, L**4** and L**5**. L**6** is a boundary used in a different embodiment, however the present embodiment uses L**5** and not L**6**. Flow f_2 can range over the interval separating line segments L**1** from L**3**, namely f_2^{MIN} and f_2^{MAX}; the range is depicted as f_2^{RANGE}. Flow f_1 can range over the interval between lines L**2** and L**4**, namely f_1^{MIN} and f_1^{MAX}, and depicted as f_1^{RANGE}. Flow f_1 can range over the interval between lines L**2** and L**4**, namely f_1^{MIN} and f_1^{MAX}, as depicted as f_1^{RANGE}. The sum of flows f_1 and f_2 is constrained to lie inside of line segment L**5** which, by construction, is always orthogonal to the cost vector C_f. Cost vector C_e is also illustrated but used in a distinct embodiment. In the present embodiment, only C_f is used. In the illustrated example of the present embodiment the constraint on total flow is set to 5, and is therefore low enough to cause L**5** to intersect L**3** and L**4**. This would not have been true had the value chosen had been 10 instead of 5. With L**5** safely out of contention, the convex hull would instead be a simple rectangle bounded by L**1** through L**4**, thereby permitting both flows to assume their respective maxima without interference. In practice operational constraints exist intrinsically or are imposed from the outside so as to ensure cost effective sharing of potentially costly network resources.

Supposing FIG. **4** to be correct, the well-known methods would select would vertex V_{3-5}, which lies at the intersection of L**3** and L**5**, or V_{4-5}, which lies at the intersection of L**4** and L**5**. These solutions, though optimal, are undesirable for the

present invention as they fail to spread available bandwidth over all channels as fairly as would a centrally located interior point of L**5**. For this reason a simple optimization method is taught, which is adapted to the particular needs of this problem and ensures a fairer allocation of constrained bandwidth among all channels.

In order to optimize use of all available bandwidth, the following general method is used, with the details illustrated in FIG. **5**. This method is a solution for the problem illustrated in FIG. **4**, which geometrically illustrates the optimization problem in the limited case of two flows, f**1** and f**2**. The following description expands the problem to an arbitrary number of clients (and therefore flows) and presents a method for solving this optimization problem.

Referring to FIG. **5**, in step **500** values are calculated for the session maxFlowRate and session.minFlowRate for each client as per the minimum and maximum constraint bound expressions in 6 and 8, respectively.

The difference between these two yields the session.flowRateRange of each client. Thus

session.flowRateRange=session maxFlowRate−session.minimumFlowRate

In step **505**, the active clients are sorted in an ascending fashion based upon their session.flowRateRange. As will be shown this critical step facilitates allocation of the remaining server bandwidth as evenly as possible among all active channels, thus maximizing the number of channels that benefit by use of the total server bandwidth. An arbitrary assignation of remaining bandwidth is likely to saturate the server before all channels have been assigned extra bandwidth, thereby favoring certain channels on an ad-hoc basis.

In step **510**, each client's channel flow rate is set to the session mimimumFlowRate.

By doing so it is ensured that the minimum flow constraint is met for each session and that the minimum flow rate is a non-increasing function of time, which is critical to the proper functioning of the CAC procedure. All clients are marked as unprocessed.

In the next step, **520**, server.flowRate is set to the sum of each active client's session.flowRate.

Next, the following is iterated over all clients in sorted sequence (during any given iteration the selected client is given by its id) by performing steps **530** through **570**. In step **530** evaluating the following expressions test for possible server saturation:

delta=(server.maxFlowRate−server.flowRate)/(qty of un-processed clients) range=client.lookup(id).session.maxFlowRate-client.lookup(id).session-.flowRate

If range is greater then delta, this implies that the server can be saturated in this iteration by allocating delta to all unprocessed clients (step **540**).

On the other hand, the 'no' path for step **530** implies that the server is not saturated and that the present client (given by id) will saturate first. Accordingly, in **550** the delta variable is set as follows:

delta=range

Next, the flow rate is incremented for all unprocessed clients by delta, causing client id to saturate.

In step **560** the server flow rate is adjusted accordingly:

server.flowRate=server.flowRate+delta*(qty of unprocessed clients)

In step **570** the client given by id, now saturated, is marked as processed.

A Method for Maximal Charge

The following relates to one embodiment for optimizing the total monetary charges within the system.

Referring back to FIG. **4**, cost vector C_c lies orthogonal to line L**6**, which intersects the convex hull at the vertex formed by the intersection of lines L**4** and L**5**, namely V_{4-5}. This cost vector, and the optimal point that it implies, favors flow f**1** over flow f**2**. In this example, this is as it should be, as the cost of service for f**1** equals 2, thus exceeding the cost of service of 1 set for f**2**. As the number of flows grows to exceed the number of distinct categories of service (and associated costs of service) the unique optimal solution, depicted in FIG. **4** for the case where every flow has a distinct cost of service, no longer applies. Once again a plurality of flows within a service category vie for bandwidth which a method should endeavor to distribute evenly. This method is derived from the previous one, and optimizes one cost category after another, starting with the most costly and ending with the least costly, or when all available capacity is allocated.

Let the service categories be denoted by k=I . . . N, where k also denotes the cost of service.

Let $C_1 \ldots C_N$ be the partition of $S_{activeClients}$ that places all clients with cost of service k in set C_k. Partition sets C_k can be ordered to form sequence SeqC=$C_N \ldots C_1$.

FIG. **6** depicts the method for implementing the method to maximize the cost of service (service charge) according to objective function 2 described above.

This method is nearly identical to the previous one. The principle difference stems from the partitioning of clients according to their category (cost) of service: clients charged most are allocated bandwidth preferentially. This is accomplished by adding another level of iteration around the method of FIG. **5**. The inner iteration (steps **650** through **680**) functions exactly as before, with the difference that its actions are limited to the clients belonging to the given service category k (i.e. C_k). This difference also holds true of step **640** which sorts category k clients according to their flow ranges prior to entry in the bandwidth-allocating inner Loop. The outer loop proceeds down a sorted sequence of service categories SeqC (generated in step **630**), starting with the category generating the greatest revenue to the service provider. Given a fairly static set of service categories, this sort need be performed only when the categories undergo change. Steps **670**, **675** and **680** are identical to their counterparts in the method of FIG. **5** (i.e. **570**, **575** and **580**).

The net effect of this method is preferential allocation of bandwidth according to category of service, and equitable treatment of clients within the same category of service.

Call Acceptance Control (CAC)

CAC for Maximal Flow

The CAC procedure applicable to this flow optimization relies on the step of accepting a new client if and only if the added load induced thereby does not compromise service to existing clients or the new one. This step could not be accomplished without the close integration with previously-described flow-modulation methods of FIGS. **5** and **6**.

According to the previous discussion, the minimum flow rate is the minimum sustained flow rate that guarantees that the associated viewer will not be subject to interruptions in service due to a shortfall of content from the media server. It follows that whenever data is being delivered at a rate in excess of the minimum flow rate, a downward adjustment toward the minimum level could be accommodated as needed to surrender bandwidth to any newcomer.

FIG. **7** depicts content flow over a channel over the course of a typical session, and also how data is delivered under

real-time streaming (D). The amount of content delivered is the same in either case, but the manner of delivery differs considerably. A session is launched at time **0** as the network is lightly loaded, and the optimizer sets an accordingly high flow rate. Another client emerges at the end of interval **700**, causing a downward adjustment to the flow rate over interval B, as available bandwidth is shared between two sessions. During both of these intervals the minimum flow rate **720** drops quickly, as data accumulates in the client's media buffers. At the end of interval B a massive influx of clients necessitates that flow be dropped to the minimum flow rate, which now lies substantially below the streaming rate D and is held until all data is delivered at the end of interval C. Note that the minimum flow rate, graphed as element **720**, diminishes monotonically over time.

The server swing capacity is defined as the difference between the maximum capacity of the server and the total minimum flow rates for all active clients. Therefore:

$$\text{swingCapacity=server.maxFlowRate}-\Sigma_{i\epsilon SactiveClients}(\text{client.lookup(i).session.minFlowRate}) \quad (14)$$

Given the monotonic decreasing nature of session minimum flow rates, server swing capacity can readily be seen to be a monotonic increasing function of time over the intervals separating client admissions, at which points it undergoes a drop as a new load is taken on. This all-important characteristic implies the following:

Any client admitted for service based on the present value of swing capacity is guaranteed to have suffident bandwidth at its disposal over the entire future course of the session.

FIG. **8** depicts the server swing capacity **800** over the course of the sessions illustrated in FIG. **7**. Swing capacity rises quickly over intervals A & B as data is delivered at high flow rates over the network. It holds steady over interval C when all channels flow at their minimum rate then jumps at the end of C before resuming its monotonic rise once again.

In this procedure FIG. **9**, which executes on a periodic basis, queued clients awaiting bandwidth are scanned in FIFO order. For each one the required bandwidth is computed as per the client's prior content selection. If the available swing capacity reduced by a safety margin exceeds the amount required then the client is activated, and swing capacity is adjusted accordingly. Otherwise two possible cases are considered: 1) under the FirstFit embodiment the procedure continues scanning clients to the end of the queue, activating clients whose requirements can be met; 2) under the FIFO embodiment, the procedure ends with the first candidate client whose requirements cannot be met.

In step **900** available server swing capacity is evaluated according to the formula

$$\text{swingcapacity=server.maxFlowRate}-\Sigma_{i\epsilon SactiveClients}(\text{client.get}(i)\text{.session.minimumFlowRate})$$

The bandwidth requirement for client id in Step **920** is obtained as follows:

required_bandwidth=client.lookup(id).contentSelection.averagePlayRate

The predicate evaluated in Step **940** is given by the expression

(required_bandwidth<=swingCapacity−server.cac_flowSafetyMargin)

In step **950**, client activation entails allocation of a session and a channel, and insertion in the set of active clients eligible for bandwidth allocation by the optimal flow modulator.

In step **960** the swing capacity is diminished by the amount reserved for the activated client:

swingCapacity=swingcapacity−required_bandwidth;

Responding to Variations in Network Capacity (Maximal Flow)

In the CAC procedure for maximal flow, a safety margin was introduced, namely server.cac_flowSafetyMargin, to provide the means for ensuring that the server's swing capacity will never fall below a minimal threshold value.

According to this procedure, the following inequality always holds true:

swingCapacity>=server.cac_flowSafetyMargin (15)

In the previous discussion a server's swing capacity provided the basis for determining whether or not a prospective client should be allocated bandwidth. In another embodiment, server swing capacity can also be interpreted as specifying the maximum amount by which the server.maxFlowRate constraint can be dropped without affecting service, should such an adjustment prove necessary due, for instance, to an influx of exogenous network traffic that diminishes the amount available for multi-media services. Parameter server.cac_flowSafetyMargin can thus be set so as to guarantee a minimum capacity to tighten the constraint on maximum server flow in response to unexpected load changes that affect the server's ability to service its existing clients as well as new ones.

Anticipating Scheduled Variations in Network Capacity (Maximal Flow)

FIG. **10** depicts how the constraint on maximum flow might be allowed to vary according to the time of day, day of the week, and so forth, in expectation of time-varying traffic flow levels extrapolated from past experience, traffic flow models, etc. Maximum flow rate **1000** can be seen to vary based upon the time of day. In practice, defining the right-hand-side of inequality constraint 1 as a time-dependent expression can impose such time-varying capacities. According to the previous description, the optimizer, which executes on a periodic basis, will automatically seek new flow levels for every active session as the constraint varies. There is, however, no guarantee that an acceptable operating point will be found for all sessions (i.e. one that respects the minimal and maximum constraints on session channel flow). One such example is the case where the server is loaded to the limit and total capacity is curtailed in excess of the aforementioned safety margin. Should such a situation arise the only recourse may well be the termination of a number of established sessions (i.e. load shedding).

The goal is to eliminate service disruptions of this sort by allowing the CAC procedure to look ahead into the future, and accept new clients only if these can be accommodated without any compromise in service in the midst of previously anticipated changes in available network capacity. The following CAC procedure generalizes the previous one: before accepting a client the test on swing capacity is repeated over a sequence of time segments that cover the proposed viewing period.

Definitions

Let

$$t\_\text{end}(i)=\text{client.lookup}(i)\text{.session.playTimeToGo}+t\_\text{now} \quad (16)$$

Let server.maxFlowRate(t) be server flow capacity as a function of time, as exemplified in FIG. **10**.

Let Seq_T(t_now)=advancing sequence of future times, lead by t_now, when server.maxFlowRate(t) undergoes

a step change. For instance, at 9:15 in FIG. **10** this sequence reads as follows: 9:15, 9:30, 11:30, 13:30, 6:30, 7:30.

The server swing capacity at a future time t is computed according to the capacity and worst-case client flows at time t.

$$\text{swingCapacity}(t)=\text{server.maxFlowRate}(t)-\Sigma_{t\_end(i)>t}(\text{client.lookup}(i)\text{.session.minFlowRate}) \quad (17)$$

It is noteworthy that the worst-case client flows at time t are expressed in terms of the present minimum flow rates, which cannot increase over time, but might hold steady. Finally, a predicate is defined that tests whether a prospective customer will cause swing capacity to be exceeded at some time t, as follows:

```
(18) boolean client__fits(i,t) {
    if(client.lookup(i).contentSelection.averagePlayRate < =
            swingCapacity(t) - server.cac__flowSafetyMargin)
        return true;
    else return false;
}
```

This procedure (FIG. **11**) is an adaptation of the first, which has been extended to consider swing capacity at times in the future when capacity undergoes scheduled changes. Before accepting a client, its minimal bandwidth requirement (which by construction of the flow modulator will never increase over time) is checked against projected swing capacity at points in time when total available capacity undergoes scheduled step changes, provided these times fall within the proposed content viewing period. A candidate is activated only if all tests succeed.

Step **1100** builds a sequence of time values (SeqT) at which step capacity changes are scheduled to occur. The first element of this sequence is t_now, representing the present.

Beyond step **1100** the queue of waiting clients is scanned in FIFO order, yielding a candidate designated by id at each iteration.

The bandwidth requirement for client id in Step **1120** is obtained as follows:

required_bandwidth=client.lookup(id).contentSelection.averagePlayRate

The worst-case end time for content flow to id is obtained according to the content selected, as follows:

*t*_end=*t*_now+client.lookup(id).selected.playTime

Steps **1130** through **1150** are executed within an iteration for each time point t in SeqT falling between t_now and t_end. This iteration is ended in step **1130** if t exceeds the time window of interest, or in step **1150** if the supply of scheduled capacity changes is exhausted.

For each time value step **1140** compares required bandwidth to projected swing capacity. Projected swing capacity at time t is:

$$\text{swingCapacity}(t)=\text{server.maxFlowRate}(t)-\Sigma_{t\_end(i)>t}(\text{client.get}(i)\ \text{session.minimumFlowRate}) \quad (17)$$

Note that only active clients whose t_end times occur after t are considered in the sum of minimum flow rates.

The predicate expression used in step **1140** at time t is thus

(required_bandwidth<=swingCapacity(*t*)−server.cac_flowSafetyMargin)

Step **1160** performs the same actions as step **660** in the previous cac flowchart

The first CAC process presented above is a special case of the present one, in which the set of step change times to server.maxFlowRate is empty (i.e. server.maxFlowRate is constant), and Seq_T(t_now)=t_now.

In preparing Seq_T(t_now), one need only consider future times that will pass before the longest possible content is played out if started at t_now. In order to sidestep problems associated with rollover (at midnight, year 2000, etc.), time is best expressed as a monotonically increasing value (e.g. seconds since Jan. 1, 1990).

CAC for Maximal Charges

The method for flow modulation presented above maximizes charges to clients with active sessions. The CAC embodiments presented previously may not be sufficient, as they do not consider the cost of service as a basis for connection acceptance. As a result they may turn away higher paying customers while granting service to lower paying ones, thereby defeating the purpose for this particular embodiment. Therefore, another embodiment is defined which offers the following features:

1. Awaiting clients are serviced in order of their respective service categories, higher paying clients first.
2. Once accepted, a client is guaranteed to receive acceptable service irrespective of its service category.
3. Under heavy load conditions higher paying customers are more likely to be accepted than lower paying ones.
4. Lower paying customers will not be starved for service when higher paying ones enjoy a surplus.
5. Available bandwidth is not thrown away needlessly while clients are being denied service.

The first objective is easily met by dividing the client queue into as many bands as there are service categories, resulting in a banded queue. Bands are ordered within the queue according to their service categories, with the costliest category in front. As prospective clients arrive and make their selection they are placed in their respective queue band according to their service category (which may be set contractually, according to content selection, etc.).

The second objective is met by employing a procedure patterned after those presented previously & offering the same guarantee. The 3rd and 4th objectives may be met by dividing total available bandwidth in as many strata as there are service categories, as depicted in FIG. **12**. Two service categories are shown, Premium and Basic, each entailing an associated cost of service. A prospective client is accepted only if there is sufficient swing capacity available within its service category. The swing capacity for a given category is given by the smaller of 1) the difference between its maximum floor flow rate—corresponding to the top of the stratum for the service category—and the sum of the minimum rates of all active sessions in its category or below, and 2) available swing capacity overall. The fifth objective is met by allowing the flow optimizer to function freely subject to its operational constraints. The imposed ceilings on call acceptance by category relate to minimum flow rates, which merely impose a floor on actual flow rates. For example, basic clients might well consume all available bandwidth (**300**) in the absence of any premium customers, yet could be throttled back toward their floor flow rates (which together cannot exceed 200 in this example) at any time should any premium customer suddenly demand service. In contrast, premium customers could consume the entire 300 bandwidth. As lower paying customers appear these would be admitted to the degree that their quota on minimum flow is not exceeded (i.e. 200) and the availability of swing capacity on the system.

The procedure according to FIG. **13** requires a number of ancillary definitions, which follow:

Let the service categories be denoted by k=1 . . . N, where k also denotes the cost of service.

Let server.maxMinFlowRate[k−1] be the top of the stratum for service category k. Note that server.maxMinFlowRate [N−1] server.maxFlowRate.

Let SK be the set of active client indices with a service category equal to or less than k. Note that S**1** is contained in S**2**, S**2** is contained in S**3**, and so forth, and that $S_N = S_{ActiveClients}$.

Let swingcapacity(k) denote available swing capacity for service category k. By construction,

$$\text{swingCapacity}(k)=\text{minimum of: }(\text{server.maxMinFlowRate}(k-1]-\Sigma_{is\ Sk}(\text{client.lookup(i).session.minFlowRate})),\ (\text{server.maxFlowRate}-\Sigma_{is\ Sactive\text{-}Clients}(\text{client.lookup(i).session.minFlowRate}))) \quad (19)$$

Now, referring to FIG. **13**:

This method is used for CAC when multiple rate tariffs are in effect, and there is a desire to maximize economic returns to the service provider while offering acceptable service to all.

All waiting clients are scanned in FIFO sequence. The actions taken in Steps **1320** and **1360** are identical to those described in connection with earlier CAC flowcharts.

Step **1340** evaluates a predicate expression that tests whether the required bandwidth can be accommodated without exceeding the lesser of 1) swing capacity available to the client's category of service, and 2) total available swing across all categories of service. The latter factor could be determinative if all available bandwidth were allocated to high paying customers, leaving lower paying ones such as the proposed client unable to draw from their unfilled quota.

Let us suppose that candidate client id belongs to rate category k.

We define the swing capacity available in rate category k as:

$$\text{swingCapacity}(k)=\text{least of:}$$

$$(\text{server.maxMinFlowRate}[k-1]-\Sigma_{isv\ Sk}$$

$$(\text{client.lookup(i).session.minimumFlowRate}))$$

and

$$(\text{server.maxFlowRate}-\Sigma_{is\ SactiveClients}$$

$$(\text{client.lookup(i).session.minimumFlowRate}))$$

The predicate expression invoked by step **1340** can now be written as follows:

$$(\text{required\_bandwidth}<=\text{swingCapacity(k)}-\text{server.cac\_flowSafetyMargin})$$

This algorithm processes queued clients in band sequence, and within every band in FIFO if the predicate evaluates to true the client is activated. Otherwise two possible cases are considered:

1) under the FirstFit embodiment the procedure continues scanning clients to the end of the banded queue, activating clients whose requirements can be met; 2) under the FIFO embodiment, the procedure ends with the first candidate client whose requirements cannot be met. Many other variations on these two embodiments might also be considered.

Anticipating Scheduled Variations in Network Capacity (Maximal Charge)

The procedure applicable to optimization of delivery charges is obtained by blending elements of the CAC method depicted in FIG. **13** into the method depicted in FIG. **11**, which applies without change. To understand how this might work it may be useful to visualize a version of FIG. **10** stratified along its length in the manner of FIG. **8**. As the maximum flow level undergoes a step change so too do the widths of its constituent strata in equal proportion.

As previously mentioned, the method of CAC described above (FIG. **11**) applies to this circumstance also, provided we alter the definition routines, (17) and (18), upon which that procedure relies, yielding (20) and (21), and adopt the banded queue organization outlined in the previous section.

The server swing capacity at a future time t is computed according to the capacity and worst-case client flows at time t.

$$\text{swingCapacity}(k,t)=\text{minimum of (}$$

$$(\text{server.maxFlowRate(t)}*(\text{server.maxMinFlowRate}[k-1]/\text{server.maxMinFlowRate}[N-1])-\Sigma_{is\ Sk\&(t\_end(1)>t)}(\text{client.lookup(i).session.minFlowRate})),$$

$$(\text{server.maxFlowRate}-\Sigma_{is\ Sactive\ Clients\&(t\_end(i)>t)}(\text{client.lookup(i).session.minFlow Rate}))) \quad (20)$$

Finally, we define a predicate that tests whether a prospective customer will cause swing capacity to be exceeded at some time t, as follows:

```
(21) boolean client_fits(i,t) {
        k = client.lookup(i).costOfService;
        if(client.lookup(i).contentSelection.averagePlayRate < =
                swingCapacity(k,t) - server.cac_flowSafetyMargin)
        return true
    else return false;
    }
```

A method for call/connection acceptance and flow modulation for network delivery of video/audio programming is thus provided. Although several embodiments have been illustrated and described, it will be apparent to those skilled in the art that various changes and modifications may be made without departing from the spirit of the invention as set forth in the following claims.

What is claimed is:

1. A method for optimal multimedia content delivery over networks from a server to one or more clients, comprising:

delineating a state variable that represents a data rate for delivery of multimedia content having a fixed duration and, wherein an initial data rate is equal to or greater than a minimum flow rate, and wherein the minimum flow rate ensures that all required content will be available to each client when needed, and a subsequent data rate, which is equal to or less than a real-time rate of play back of the multimedia content;

delineating a set of conditions which represent time-varying constraints on the data rate of said multimedia content said conditions including:

(1) the total data rate for all clients does not exceed the maximum throughput of the server or network, whichever is least;

(2) the data rate from server to client does not exceed the maximum data rate for the client;

(3) the data rate of the client will never overflow a client buffer;

(4) the server will never underflow; and

(5) the data rate from the server will never be less than the client's minimum data rate, and wherein the minimum data rate is a non-increasing function of time obtained by dividing the content not yet delivered by the remaining play time;

delineating a cost function which represents the value of a proposed solution;

performing periodic computations in compliance with conditions (1)-(5) to obtain a state value that maximizes said cost function; and

periodically adjusting the data rate to each client to maintain an optimal solution over a given period of time.

2. A method as in claim **1**, wherein said conditions further include

(6) the current maximum client data rate is given by the minimum of:

the stored initial maximum client data rate;

the data rate required to fill the remaining client buffer during the current of said periodic computations;

the data rate required to complete the delivery of said multimedia content;

the client data rate never exceeds said current maximum client data rate; and

whereby said current maximum client data rate is periodically recomputed to maintain an optimal solution over a given period of time.

3. A method as in claim **2**, wherein:

said cost function represents maximal throughput and is given by the sum of said client data rates for all active clients.

4. A method as in claim **2**, wherein:

said cost function represents maximal charge and is given by the sum for all active clients of said client data rates times the client's cost of service.

5. A method as in claim **3** for bandwidth allocation for delivery of multimedia data from server to one or more clients over a network, comprising the steps of:

determining the maximum flow rate and minimum flow rate for each client;

determining the flow rate range for each client as given by the difference between said maximum flow rate and said minimum flow rate;

initializing current flow rate for each client as said minimum flow rate and summing said flow rate into total server flow rate; and

allocating remaining server bandwidth to remaining clients until they each saturate or no bandwidth remains.

6. A method as in claim **5** wherein said step of allocating remaining server bandwidth to remaining clients comprising:

sorting the list of clients according to said flow rate range;

determining equally-allocated remaining server bandwidth if allocated evenly to all remaining unprocessed clients;

determining the range of remaining client bandwidth as given by the difference between said maximum flow rate and said minimum flow rate;

determining saturation by comparing said equally-allocated remaining server bandwidth and said range of remaining client bandwidth, and allocating the lesser of these two amounts to each remaining client flow rate; and

whereby allocating flow to remaining clients based upon the sorted client range flow rates and determining allocation of remaining server bandwidth based upon a comparison of saturation of server versus saturation of each client maximizes allocation of total bandwidth for maximal flow rate to maximum number of clients.

7. A method as in claim **4** for bandwidth allocation for delivery of multimedia data from server to one or more clients over a network, comprising the steps of:

determining the maximum flow rate and minimum flow rate for each client;

determining the flow rate range for each client as given by the difference between said maximum flow rate and said minimum flow rate;

sorting the list of clients according to said flow rate range;

initializing current flow rate for each client as said minimum flow rate and summing said flow rate into total server flow rate; and

allocating remaining server bandwidth to remaining clients such that lower paying clients receive bandwidth only if higher paying ones are saturated.

8. A method as in claim **7** wherein said step of allocating remaining server bandwidth to remaining clients comprises the steps of:

for each remaining unprocessed client:

determining equally-allocated remaining server bandwidth if allocated evenly to all remaining unprocessed clients;

determining the range of remaining client bandwidth as given by the difference between said maximum flow rate and said minimum flow rate;

determining saturation by comparing said equally-allocated remaining server bandwidth and said range of remaining client bandwidth, and allocating the lesser of these two amounts to each remaining client flow rate; and

whereby allocating flow to remaining clients based upon the sorted client range flow rates and determining allocation of remaining server bandwidth based upon a comparison of saturation of server versus saturation of each client maximizes allocation of total bandwidth for maximal flow rate to maximum number of clients.

9. A method for connection acceptance control for delivery of multimedia data from server to one or more clients over a network, comprising the steps of:

determining server swing capacity given by the difference between the total server bandwidth and the sum of the minimum flow rates of all currently-connected clients receiving multimedia data having a fixed duration, wherein the minimum flow rate for each client is expressed as a non-increasing function of time obtained by dividing data not vet delivered by remaining play time, and wherein the minimum flow rate ensures that all required data will be available to each client when needed; and

allocating server bandwidth for each prospective client which will fit without server bandwidth saturation, as determined by comparing an average data play rate of each prospective client with the remaining bandwidth, represented by said server swing capacity, available to the server.

10. A method as in claim **9** wherein said remaining bandwidth available to the server is given by said server swing capacity.

11. A method as in claim **10** wherein said remaining bandwidth available to the server is give by said server swing capacity less a server flow safety margin, thereby allowing server capacity to be subsequently lowered by up to the safety margin without requiring load shedding, and without affecting client sessions in process.

12. A method as in claim **9** wherein said step of allocating server bandwidth for each prospective client which will fit without server bandwidth saturation comprises:

allocating server bandwidth to each prospective client sequentially until a prospective client is located in which said average data play rate exceeds said server swing capacity.

13. A method as in claim **9** wherein said step of allocating server bandwidth for each client which will fit without server bandwidth saturation comprises:

allocating server bandwidth to each prospective client sequentially for each client which can be activated without server bandwidth saturation.

14. A method for bandwidth allocation for delivery of multimedia data from a server to one or more clients over a network, the method comprising:

storing multimedia data on at least one server, the multimedia data having a fixed duration;

delivering the multimedia data from the at least one server to at least one client device upon demand of the at least one client by way of a network having a defined bandwidth, and wherein the multimedia data is available for playback upon client request;

storing a sequence of data representing scheduled bandwidth changes for the at least one server;

determining a maximum flow rate and a minimum flow rate for each of the one or more clients at the present time, the determination of the minimum flow rate being based on a non-increasing function of time obtained by dividing multimedia data not yet delivered by remaining play time, and wherein the minimum allowed flow rate ensures that all required multimedia data will be available to each client when needed;

determining the flow rate range for each client as given by the difference between said maximum flow rate and said minimum flow rate;

sorting the list of clients according to said flow rate range;

initializing current flow rate for each client as said minimum flow rate and summing said flow rate into total server flow rate; and

allocating remaining server bandwidth to remaining clients.

15. The method of claim **1**, wherein the data rate ensures that all required content will be available to each client when needed.

16. The method of claim **1**, further comprising ceasing delivery of the multimedia content to the at least one client when the content not yet delivered is equal to zero.

17. The method of claim **1**, further comprising accepting a new client by:

determining an admission capacity of the bandwidth;

admitting a prospective client if the clients minimum allowed value of the state variable is less than the admission capacity; and

wherein a client admitted for service is guaranteed to have sufficient content flow over the entire session.

18. The method of claim **14**, further comprising ceasing delivery of the multimedia data to the at least one client when the data not yet delivered is equal to zero.

19. The method of claim **14**, further comprising accepting a new client by:

determining an admission capacity of the bandwidth;

admitting a prospective client if the remaining clients minimum allowed value of the state variable is less than the admission capacity; and

wherein a new client admitted for service is guaranteed to have sufficient data flow over the entire session.

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